高等学校专业英语教材
上海市普通高校优秀教材

通信工程专业实用英语

（第 2 版）

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内 容 简 介

本书为“上海市普通高校优秀教材”。

相比普通大学英语，专业英语以培养学生的职业岗位综合能力为目标，根据相关行业的发展趋势和就业需求，有针对性地对学生进行职业技能培养。

本书在第一版教材基础上，注重实际、强调应用，结合专业技术变迁和发展进行内容的增删和调整，形成了“基本通信概念与系统”、“常见通信业务”、“最新通信技术及其应用”三个部分，共 16 个单元。

本书尽量结合实际通信系统的原理与技术进行编写，并在每单元附有 1～5 篇英文行业资讯作为阅读材料，以及形式多样的课后练习和答案，可以帮助读者有效学习和自我拓展。既可作为本科电子与通信类专业英语的教学用书，也可作为计算机通信、网络类专业相关工程技术人员参考用书。

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前　　言

随着信息、通信技术的飞速发展，电子、通信专业引进了大量国外先进的行业相关标准、技术与设备，对通信从业者的专业英语阅读、理解能力要求越来越高，通信专业英语教学的重要性也日趋明显。

专业英语以大学英语为基础，但在词汇、语法及文风上带有浓厚的专业特色，相比普通大学英语，专业英语更注重培养学生的职业岗位综合能力。目前国内本科通信专业英语教材在选题上普遍倾向计算机（网络）通信方向，侧重于介绍通信网络的结构、类型、组成原理和协议等，与实际通信技术专业的基础理论和行业热点联系松泛，不利于学以致用。

本教材紧密围绕电子通信专业实际应用技术和最新行业资讯，由浅入深，精心组织、编撰，受到读者普遍好评，第一版经几年的使用，被评为“上海市普通高校优秀教材”。

本版教材在第一版教材基础上，继续突出“注重实际，强调应用”这一特点，结合本专业领域近年来的技术变迁和发展，进行了增删、调整和改编，形成了“基本通信概念与系统”、“常见通信业务”、“最新通信技术及其应用”三个部分，具有如下特色：

1．“基本通信概念与系统”部分由第 1～7 单元、第 15 单元构成，是将第一版中有关基础理论、技术的单元进行了合并、精简，形成。内容包括现代通信的基本概念和技术，如通信系统的组成、通信频段划分、线性/非线性调制理论，编码技术、多址接入技术、带限信道的信号传输、扩频调制、2G 移动通信技术、电信增值服务等。

2．“常见通信业务”部分由第 8～14 单元构成，主要介绍目前通信行业的最新主流技术及其应用，包括 3G/4G 移动通信系统、电路交换/分组交换、光纤传输技术和系统、物联网技术、VOIP 技术、嵌入式技术等。

3．“最新通信技术及其应用”部分主要针对的是目前大热的汽车电子技术，本书在 16 单元针对其相关概念和应用进行了介绍，以期为未来在该领域就业的同学进行基础铺垫。

4．针对学生理论基础水平参差不齐的状况，第二版教材依然在每单元附有 1～5 篇英文行业资讯，以便教师根据情况有选择地组织教学拓展。

5．此外，该书还附有形式多样的课后练习以及免费的电子课件和习题答案。一方面可以帮助读者有效地复习课程中所学的内容；另一方面也便于学有余力的学生在课后进行自我拓展训练。

本教材参考学时为 64～80 学时，是通信、电子类教学用书，同时也可作为计算机通信、网络类专业相关工程技术人员的参考用书。

本书由上海师范大学天华学院陶亚雄教授主编、徐振副主编，天津师范大学刘南平教授主审，上海师范大学天华学院工学院的朱国权、刘伟、赵兰、徐会彬、王永明老师，以及重庆电子工程职业学院通信工程学院的黄祎、曾晓宏、刘良华、刘之舟老师参与了其中部分单元的编写、校对及资料收集等工作。

该书在编写过程中得到了各位参编老师及上师大天华学院领导的大力支持和帮助，在此表示衷心的感谢；同时也对提供文献参考资料的专家、学者表示深深的谢意。

由于水平、精力有限，书中疏漏甚至错误在所难免，欢迎各位读者批评指正。

编　者

20013 年 11 月

目　　录

Unit 1　Brief Introduction of Modern Communication

1.1　Text

1.1.1　Communication

Modern communication means a technology using light wave and electromagnetic wave to transmit or exchange information from one place to another rapidly and accurately, so it's also called ***telecommunication*** technique.

Along with the unceasing development and fusion of communication technique, computer technique and control technology, performance of communication systems have enormously expanded, such as visible text, electronic mail-box, video telephone and conference, etc., accompanied with the communicating content extension from simplex voice and text signals to multimedia information including sound, text, data, picture and so on. Not only efficient information transmission, but also information collecting, processing, storage and displaying are carried out by modern communication network.

Classification of modern communication systems is different along with the different classifying manners.

1. Simplex & Half-duplex & Full-duplex Communication

According to the information direction transmitted in channel, modern communication systems can be divided into the ***simplex communication*** systems, ***half-duplex communication*** systems, and ***full-duplex communication*** systems.

In simplex communication systems such as radio and television broadcasting, signals can only flow in one direction. In half-duplex communication systems, signals can flow in both directions, but only one direction at a time (not simultaneously). Typically, once a party begins to receive a signal, it must wait for the transmitter to stop transmitting, before replying. Full-duplex systems are employed in many communication networks, in which signals can flow in both directions.

2. Serial & Parallel Communication

According to the number of information communicating approaches, modern communication systems can be divided into the ***serial communication*** systems and the ***parallel communication*** systems.

Serial transmission is the process of sending data one bit at one time, sequentially, over a communication channel, as shown in Fig 1-1(a). Parallel transmission is mainly employed in real-time communication and data communication between computer and its peripherals, in which several data bits are packed together and transmitted simultaneously as shown in Figure.1-1(b).

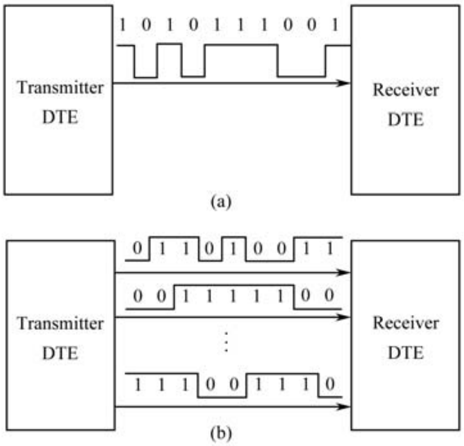


Figure 1-1 Serial communication and Parallel communication

3. Synchronous & Asynchronous Communication

According to the control methods of information transmitted in channels, modern communication systems can be divided into the ***synchronous communication*** systems and the ***asynchronous communication*** systems.

In asynchronous communication system, every symbol is transmitted independently at variable data rate, only one symbol at one time. A start bit (e.g. logic level 1) serves to represent the start of a new symbol, and a stop bit (e.g. logic level 0) serves to the represent the end of a symbol. Usually, the start bit length takes one bit while the stop bit length required by the system can be 1, 1.5 or 2 bits as shown in Figure1-2. Since the transmission of every symbol usually requires 2~3additional bits, asynchronous transmission usually lacks efficiency.

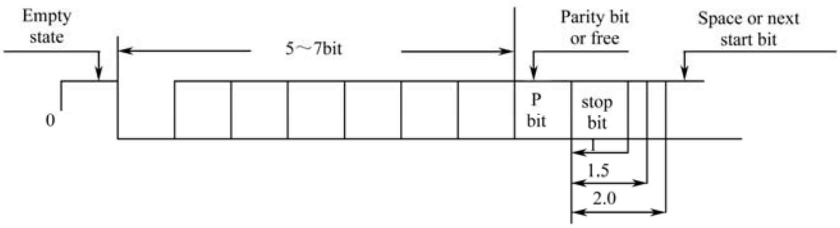


Figure 1-2 Asynchronous communication

In synchronous communication system, information is transmitted in forms of data block. Each block has a preamble bit and postamble bit respectively for symbolizing the start and end of block. Apparently, synchronous communication system is more efficient than asynchronous communication system, and is more adaptive for high speed data transferring .

4. Point-to-point & one-to-multi-points & multi-to-multi-points Communication

According to the line connecting modes and signal interacting ways between signal source and destination, modern communication systems can be divided into ***point-to-point communication***

systems, ***one-to-multi-points communication systems*** and ***multi-to-multi-points communication*** systems.

In point-to-point communication system shown in Figure1-3(a), the connection between the terminals, such as terminal A and B, was generally implemented through a dedicated line. In point-to-multipoint communication system shown in Figure1-3(b), connection between every terminals (such as terminal A, B,…, et al.) is accomplished via a transferring equipment. In multipoint-to-multipoint communication system, data is transmitted flexibly between several terminals through a switching device, with the direct or stored-and-transferred method.

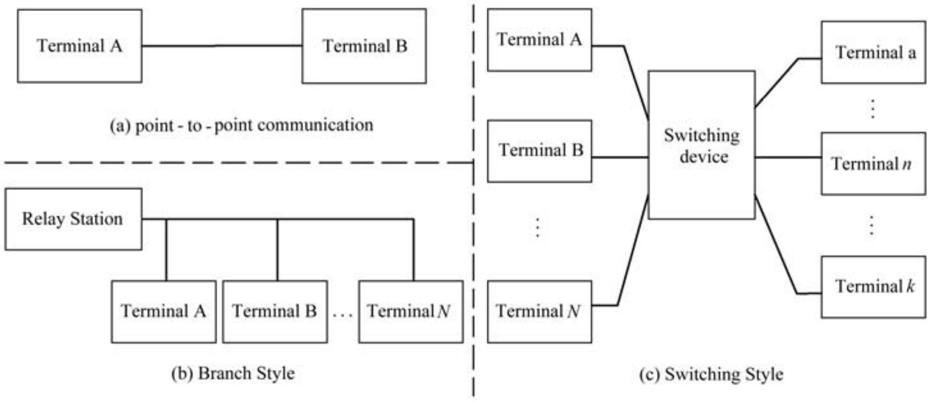


Figure 1-3 point & one-to-multi & multi-to-multi points communication

1.1.2 History of Modern Communication

Communication comes up along with the history of humanity since people have to transmit information and exchange their views each other. Since the invention of electric battery by Alessandro Volta in 1799, people had begun to try to communicate making use of electricity. The development of modern communication can be illustrated by those milestone events listed in the following Table 1-1.

Table 1-1 Memorabilia of modern communication

Age	Event	Significance
1837	Samuel Morse invented the electric telegraph	Beginning of a new era that electricity being used by people for long-distance information transmission
1876	A.G.Bell invented telephone	Transmitting voice signals by using current intensity directly
1864，1887	Maxwell predicted the existence of electromagnetic radiation in 1864, and was verified experimentally by Hertz it in 1887	Providing modern wireless communication with theory basis
Early 20th century	Amplitude modulation (AM) appeared	Changing communication signal from simplex audio signal to hybrid signal of voice, music, picture signals
1933	Frequency modulation (FM) appeared	Improved communicating quality by overcoming the bug that AM signal is prone to interference, and impelled the development of mobile communication

（续）

Age	Event	Significance
1928 1937	***Nyquist's Theorem*** was proposed; A.H.Reeves invented pulse code modulation (PCM) communication technique	Development of communication from analog to digital transmission; Analog signal being digitally transmitted via PCM technique, and improving the ability of communication system to anti-jamming
1940s-1950s	Shannon Formula, ***Non-distortion Coding Principle***, ***Error-correction Coding Principle***, Signal and Noise Theory, Modulation Principle, Signal Detection Theory appeared	Providing communication validity and reliability with theory basis, promoting communication technology to be mature and progressive
l960	The first satellite for communication launched successfully	Breaking the new path for international communication, bringing on the rapid development of space communication
1960s	***Cable television***, ***laser communication***, radar, computer network and digital communication technology appeared	***Photoelectricity processing technology*** and ***radio astronomy*** getting great development
1970s	***large-scale-integrated circuit*** (***LSI***), Private (Automatic) Branch exchange, microprocessor developed rapidly	Commercial ***satellite communication***, optical fiber communication getting rapid development
1980s	***Very-large-scale-integrated circuit*** (***VLSI***), Integrated Services Digital Network (ISDN) appeared	Mobile communication and optical fiber communications coming into application

Technical words and phrases

communication [kəmju:ni'keiʃən] *n.* 通信；联络
electromagnetic [i'lektrəumæg'netik] *adj.* 电磁的；电磁学的
telecommunication ['teli-kəmju(:)ni'keiʃən] *n.* 电信，无线电通信；电信学
unceasing [ʌn'si:siŋ] *adj.* 不停的，持续的
visible ['vizəbl] *adj.* 看得见的；明显的，显而易见的
video [vidiəu] *adj.* 视频的；录像的
telephone ['telifəun] *n.* 电话；电话机
channel ['tʃænl] *n.* 信道，频道
simplex ['simpleks] *adj.* 单纯的，单一的
simplex communication 单工通信
half-duplex *n.* [计]半双工
half-duplex communication 半双工通信
duplex ['dju:pleks] *adj.* 双倍的，复式的，[电信、计]双工的，双向的
full-duplex communication 全双工通信
serial ['siəriəl] *adj.* 连续的；系列的；按顺序的
serial-communication 串行通信
parallel [pærəlel] *adj.* 平行的，相同的，类似的，并联的
parallel-communication 并行通信
synchronous ['siŋkrənəs] *adj.* 同时发生的；同步的
asynchronous [ei'siŋkrənəs] *adj.* 不同时的；[电]异步的
signal ['signəl] *n.* 信号
source [sɔ:s] *n.* 来源，水源；消息来源

point-to-point communication　点到点通信
one-to-multi-points communication　点到多点通信
multi-to-multi-points communication　多点到多点通信
Amplitude modulation (AM)　幅度调制，调幅（常规双边带调幅）
Frequency modulation (FM)　频率调制，调频
Nyquist's Theorem　奈奎斯特定理
pulse code modulation (PCM)　脉冲编码调制
Shannon Formula　香农公式
Non-distortion Coding Principle　不失真编码原理
Error-correction Coding Principle　纠错编码原理
Signal and Noise Theory　信号和噪声理论
Modulation Principle　调制原理
Signal Detection Theory 信号检测理论
Very-large-scale-integrated circuit (VLSI)　超大规模集成电路
large-scale-integrated circuit (LSI) 大规模集成电路
Integrated Services Digital Network (ISDN)　综合业务数字网
optical fiber communications　光（纤）通信
photoelectricity　[fəutəuilek'trisiti]　[物]光电(学)；光电现象
PBX (Private Branch Exchange)　专用分局交换机
PABX (Private Automatic Branch eXchange)　自动用户小交换机
microprocessor　微处理器
principle　['prinsəpl]　*n.*　法则，规则，原则；原理，定理
satellite communication　卫星通信
space communication　宇宙通信，空间通信

1.2　Reading Materials

1．Google Wins Internet Advertising Contract with China Telecom

Search giant, Google Inc., is poised to increase its share of the Chinese internet advertising market, due to a new agreement with China Telecom.

Google has won the right to place ads of 400 of the telecom giant's websites, giving it valuable leverage against Microsoft, and leading local rival, Baiducom.

"This is a big win for Google because Microsoft and Baidu both wanted this agreement with China Telecom", said Analysis International internet analyst, Foo Xinghua, in a telephone interview. "China Telecom likely picked Google because they have better technology for Web ads."

China's internet advertising market will surge to US$3.1 billion in 2011, from $420 million in 2005, according to estimates.

（Source: teleclick）

2．Google, China Telecom Form Online Ad Alliance

Google has inked an agreement with China Telecom to sell online advertising on 400 of its web sites. This venture will allow Google into a new market space, as it will provide Google with entry into China Telecom's network of web sites that reached out to a domestic audience.

China Telecom has revealed that its subscriber base touches 30.5 million broadband users. Under the terms of the agreement the companies will share the revenue generated, though the specific financial details of the deal were not divulged.

This is the third time both the companies are coming together to do business. Google wants to tap the burgeoning Chinese market.

（Source: sda-india）

3．China to Launch Mobile Phone TV Satellites in 2008

China is going to launch two satellites for mobile multimedia broadcasting in May 2008, revealed an expert involved in the formulation of China Mobile Multimedia Broadcasting (CMMB) system, the recommended industrial standard announced by the country's broadcasting regulator in late October.

The commissioning of the satellite system is considered to be a significant step for the operation of China's independently-developed mobile multimedia broadcasting system, as the country plans to build a CMMB network with large-scale satellite signal as a major mode of signal coverage and the transmission on the ground as a complement, in view of the country's vast extension of territories with different development stages.

Remarkably, the nation's broadcasting administration will adopt the experience of the telecom department in building the network for the next-generation telecommunications in the CMMB network project.

It will start to build a ground test network by the end of this year, and complete the test network in the middle of next year and by then start system testing; to complete the building of the ground network for commercial use and start commercial operation test by the end of next year; and will form a nationwide CMMB network with the commissioning of the satellite system in the first half of 2008 and by then officially start providing mobile multimedia broadcasting services, before the opening of Beijing Olympics.

(Source: stocknews.com.cn)

4．Asia Pacific region plans for next-generation networks (NGN)

Geneva, 12 April 2007—ITU and the Asia Pacific Telecommunity (APT) jointly organized a workshop in Bangkok, Thailand to plan for the implementation and development of Next-Generation Networks (NGN) in the region. Over 180 experts from 24 countries representing APT and ITU Members, international organizations and the private sector joined the forum, which was inaugurated by Mr Kraisorn Pornsutee, Permanent Secretary, Ministry of Information and Communication Technology, Royal Government of Thailand.

NGN is a catch phrase for the network infrastructure that will enable advanced new services offered by mobile and fixed network operators in the future, while continuing to support all existing services. This next-generation architecture will help leverage new technologies to dramatically reduce the cost of market entry, increase flexibility and accommodate seamlessly in a single multiservice network both voice and data.

Mr Malcolm Johnson, Director of the ITU Telecommunication Standardization Bureau, said, “NGN has the potential to accelerate the deployment of telecommunication networks and services in developing countries.”As cost and revenue are the drivers of this development, the capital cost of deploying NGN technology, both in the core of the network, and the operating costs are significantly lower than circuit switched technologies. “This will enable rapid expansion of network capabilities,” Mr Johnson added. “NGN will also enable a range of multimedia services to be provided more easily and with less cost, and so increase potential revenues. It offers the opportunity for developing countries to leapfrog several generations of technology.” He also stressed the importance of “bridging the standardization gap” by planning for NGN at regional levels.

5．International steps taken to build global Information Society

Geneva, 20 July 2006—Implementation of the outcomes of the recently concluded World Summit on the Information Society (WSIS) gathered momentum with the launch of the United Nations Group on the Information Society (UNGIS). High level representatives of twenty-two UN agencies met on Friday, 14 July 2006 at ITU Headquarters in Geneva under the chairmanship of ITU Secretary-General Yoshio Utsumi to facilitate the process.

UNGIS will serve as an interagency coordinating mechanism within the UN system to implement the outcomes of WSIS. The Group will enable synergies aimed at resolving substantive and policy issues, avoiding redundancies and enhancing effectiveness of the system while raising public awareness about the goals and objectives of the global Information Society. UNGIS will also work to highlight the importance of ICTs in meeting the Millennium Development Goals.

To maximize its efficiency, the Group agreed on a work plan in which it would concentrate its collective efforts each year on one or two cross-cutting themes and on a few selected countries.

1.3 Exercises

1. Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）单工通信系统
（2）全双工通信
（3）同步通信
（4）异步通信
（5）点到点通信
（6）串行通信
（7）并行通信

（8）电信
（9）有线电视
（10）光电处理技术
（11）射电天文学
（12）卫星通信
（13）大规模集成电路
（14）超大规模集成电路
（15）奈奎斯特定理
（16）无失真编码理论
（17）纠错编码

2. Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）Modern communication means a technology using light wave and（　　）wave to（　　）or exchange information from one place to another rapidly and accurately, so it's also called（　　）technique.

（2）According to the information direction transmitted in channel, modern communication systems can be divided into the（　　）communication systems, half-duplex communication systems, and（　　）communication systems.

（3）According to the number of information communicating approaches, modern communication systems can be divided into the（　　）communication systems and（　　）communication systems.

（4）According to the control methods of information transmitted in channels, modern communication systems can be divided into the（　　）communication systems and（　　）communication systems.

（5）Modern communication can be divided into analog communication and（　　）communication according to the transmitted signal type. If signals transmitted are（　　）signals, then the communication system is an analog communication system.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（　　）Because of using electromagnetic wave to transmit or exchange information from one place to another rapidly and accurately, modern communication is also called telecommunication technique.

（2）（　　）Classification of modern communication systems is different along with the different classifying manners.

（3）（　　）According to the information transmitting direction, modern communication systems can be divided into point-to-point communication systems, one-to-multi-points communication systems and multi-to-multi-points communication systems.

（4）（　　）Morse's invention of line telegraphy began a new era of light-wave communication.

（5）（　　）Hertz's prediction and Maxwell's demonstration about the existence of electromagnetic radiation provided modern wireless communication with theory basis.

（6）（　　）Frequency modulation improved communicating quality and impelled the development of mobile communication.

（7）（　　）The successful launch of first communication satellite broke the new path of space communication.

（8）（　　）SLSI and ISDN technologies promoted mobile communication and optical fiber communications into application.

1.4　课文参考译文　现代通信简介

1.4.1　通信

现代通信（communication）指利用光、电技术手段，借助光波或电磁波，实现从一地向另一地迅速而准确地信息传递和交换的技术，也称之为电信（telecommunication）技术。

通信技术、计算机技术和控制技术不断发展与融合极大地扩展了通信的功能，通信传递的内容也已从单一的语音或文字转换为集声音、文字、数据、图像等的多媒体信息，通信网不仅能有效地传递信息，还可以存储、处理、采集及显示信息，实现了可视图文、电子信箱、可视电话、会议电视等多种信息业务功能。

按照不同的划分依据，通信系统有多种不同的分类。

1. 单工、半双工、双工通信

按信息在信道中的传输方向，通信方式可分为单工通信、半双工通信和全双工通信。单工通信系统的信号只能单方向传送，如广播、电视系统传输系统。半双工通信系统的信号可以在两个方向上传输，但时间上不能重叠，即通信双方不能同时既发送信号又接收信号而只能交替进行。全双工通信方式是目前使用最多的通信方式，其信道可以随时双向传输信息。

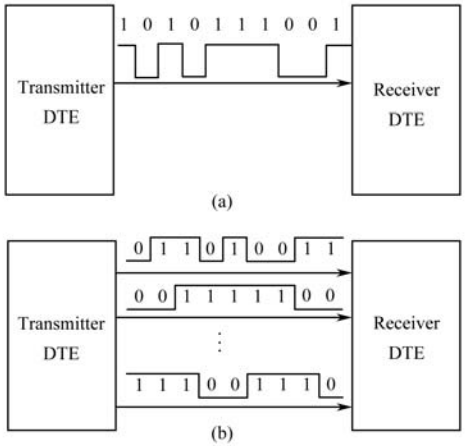
2. 串行、并行通信

按照通信双方传输信息的路数，通信方式又可分为串行通信和并行通信。串行传输的数据码元是一位接一位地在同一条信道上传输的，如图 1-1（a）所示。并行传输常用于现场通信或计算机与外设之间的数据传输，一次将构成一个字符的多个码元同时传送，如图 1-1（b）所示。

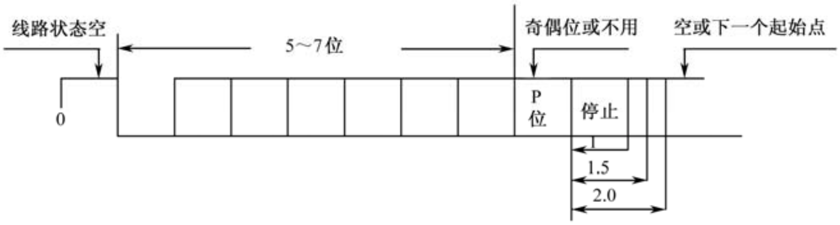
3. 同步、异步通信

按照信息在信道中传输的控制方式，通行的方式可分为同步传输和异步传输。

异步通信系统的每个字符都以不均匀速率独立发送，每次只传送一个字符，分别由起始位（如逻辑电平 1）、停止位（如逻辑电平 0）表示一个新字符的开始和结束。起始位一般占一位码元时间，停止位可根据需要取 1、1.5 或 2 位码元宽度，如图 1-2 所示。由于每个码元的传输都需要增加 2～3 比特的附加信息，异步通信传输效率较低。



译图 1-1　串行通信和并行通信



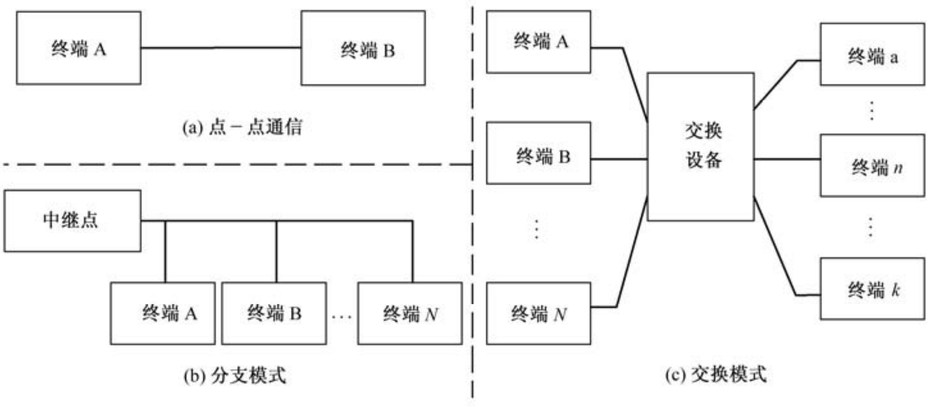
译图 1-2　异步通信

同步通信以数据块为单位传输信息，并在每个数据块的前、后端分别加上前文（preamble）、后文（postamble），以表示数据块的开始、结束。显然，同步通信的效率要比异步通信的效率高，更适用于高速数据传输的场合。

4. 点—点、点—多点、多点—多点通信

根据信源、信宿之间不同的线路连接与信号交互方式，通行又可以分为点到点的通信、点到多点通信以及多点到多点的通信等。

点到点的通信方式如图 1-3（a）所示，进行通信的两个终端 A 与 B 之间通过专用线路直接进行信息交流。点到多点的通信方式中，每个终端（如终端 A，终端 B，…，终端 *N*



译图 1-3　点–点、点–多点、多点–多点通信

等）都经过同一个信道与转接站连接，其相互之间的通信必须经过转接站转接才能实现。多点到多点通信方式中，借助交换设备，各个终端之间可以灵活地采用直接连通线路的方式，或存储、转发的形式进行通信。

1.4.2 现代通信简史

人类必须要进行思想交流和信息传递，所以有人类就有通信。从 1799 年伏特发明电池以来，人们就开始努力试图利用电来进行通信了。其发展过程可以从表 1-1 所列事件代表的阶段得以描述。

译表 1-1 现代通信大事记

年代	事件	意义
1837	莫尔斯（Morse）发明了电报	人类开始利用电进行远距离消息传递
1876	贝尔（A.G.Bell）发明了电话机	直接利用电流强弱传送语音信号
1864，1887	麦克斯韦（Maxwell）预言电磁辐射，赫兹（Hertz）实验证实	为现代的无线电通信提供了理论根据
20 世纪初	出现了幅度调制 AM	将通信内容由单一的语音传送变为语音、音乐、图像等多种信号传送
1933	调频 FM 技术出现	FM 技术克服了 AM 技术容易受到干扰的缺点，改善了通信质量，推动了移动通信技术发展
1928 1937	奈奎斯特（Nyquist）定理被提出； 瑞维斯（A.H.Reeves）发明 PCM（脉冲编码调制）通信	通信技术由频分复用发展到时分复用，开始由模拟通信转向数字通信； 通过 PCM 技术，模拟信号被数字化传送，进一步提高了抗干扰能力
20 世纪 40～50 年代	香农公式和不失真编码原理、纠错编码原理、信号和噪声理论、调制原理以及信号检测理论出现	对通信的有效性和可靠性提供了理论依据，促进了通信技术的成熟与进步
l960	第一颗通信卫星发射成功	开辟国际通信通道，促进空间通信发展
20 世纪 60 年代	电缆电视、激光通信、雷达、计算机网络和数字技术出现	光电处理技术和射电天文学飞速发展
20 世纪 70 年代	大规模集成电路、程控数字交换机、微处理机迅猛发展	商用卫星通信、光纤通信迅猛发展
20 世纪 80 年代	超大规模集成电路，综合业务数字网出现	移动通信、光纤通信得到应用

Unit 2 Modulation Techniques

2.1 Text

2.1.1 Digital & Analog Communication

Modern communication can be divided into analog communication and digital communication according to the type of transmitted signal. If signals transmitted in a communication system are analog signals, the communication system is an ***analog communication system.*** On the other hand, a ***digital communication system*** is a system which transmits and processes digital signals.

In conventional telecommunication system, signals transmitted in channel change with the voice of user during the whole talking period. These signals, shown in Figure 2-1 (a), are continuous in time and amplitude, so these signals are called the ***analog signals***. Unlike analog signal, digital signal is a kind of signal which is discrete both in time and amplitude. The most widely encountered digital signal is the one which has only two amplitudes—the 0 and 1, and is usually called binary digital signal shown in Figure 2-1 (b). Generally, digital signals can de described as binary and M-ary (M >2) signals, clearly, the M-ary signal means a type of digital signal which has M amplitudes for selection. Figure 2-1 (c) illustrates a 4-ary digital signal with 0,1,2,3 etc. amplitudes to choose.

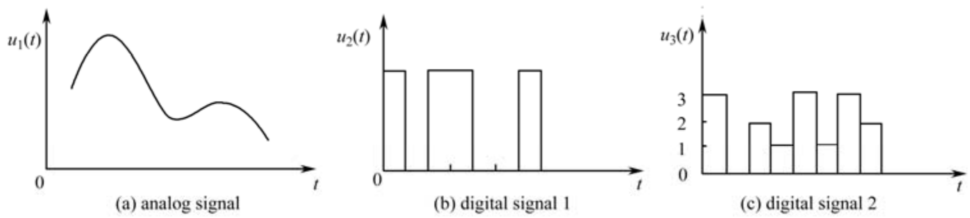


Figure 2-1 analog signal and digital signals

Comparing with a digital signal, the bandwidth transmitting an analog signal is relatively narrow, and the efficiency of channel bandwidth usage for analog communication system is higher than that of the digital communication system. It's difficult for analog signals to distinguish the noise from original signals, and the capability of anti-interference in analog communication system is just passable.

The majority of modern communication systems are digital communication systems, so the analog signals have to be converted into corresponding digital signals before they are processed and transmitted in digital communication systems. The converting process is called ***Analog / Digital conversion,*** which usually contains the following three steps.

① Sampling: transforming analog signal into the time-discrete and amplitude-continuous signal;

② Quantizing: transforming the sampled signal into fully discrete signal;

③ Coding: using binary code words to describe the quantized digit.

The process of receiving signals in the receiving end of a digital communication system is completely the opposite to the process in sending end, which means that the receiving signal must have to be converted back into analog signals by a ***Digital/Analog conversion***.

2.1.2 Elements of a Digital Communication System

Figure 2-2 illustrates the functional diagram and the basic elements of a digital communication system. The source output may be either an analog signal, such as audio signal, or a digital signal, such as the output of a computer, which is discrete in time and has a finite number of output characters. In a digital communication system, the messages produced by the source are converted into a sequence of binary digits. Ideally, we should like to represent the source output (message) by as few binary digits as possible. In other words, we seek an efficient representation of the source output that results in little or no redundancy. The process of efficiently converting the output of either an analog or digital source into a sequence of binary digits is called source encoding or data compression.

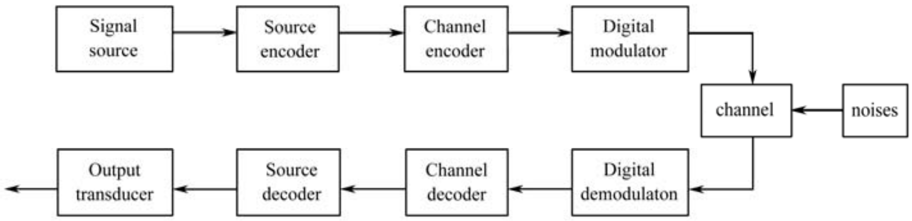


Figure 2-2 Functional diagram of a digital communication system

The sequence of binary digits from the source encoder, which is called the information sequence, is passed to the channel encoder. The channel encoder introduces, in a controlled manner, some redundancy in the binary information sequence, which can be used at the receiver to overcome the effects of noise and interference encountered in the transmission of the signal through the channel. Thus, the added redundancy serves to increase the reliability of the received data and improve the fidelity of the received signal. In effect, redundancy in the information sequence aids the receiver in decoding the desired information sequence. More sophisticated encoding involves taking k information bits at a time and mapping each k-bit sequence into a unique n-bit sequence, called a code word. The amount of redundancy introduced by encoding the data in this manner is measured by the ratio n/k. The reciprocal of this ratio, namely k/n, is called ***the rate of the code*** or, simply, ***the code rate***.

The binary sequence at the output of the channel encoder is passed to the digital modulator, which serves as the interface to the communications channel. Since nearly all of the communication channels encountered in practice are capable of transmitting electrical signals, the primary purpose of the digital modulator is to map the binary information sequence into signal waveforms.

At the receiving end of a digital communication system, the demodulator processes the channel-corrupted transmitted waveform and reduces the waveforms to a sequence of numbers that

present estimates of the transmitted data symbol (binary or M-ary). This sequence of numbers is passed to the channel decoder, which attempts to reconstruct the original information sequence from knowledge of the code used by the channel encoder and the redundancy contained in the received data.

A measure of how the demodulator and decoder perform is the frequency with which errors occur in the decoded sequence. More precisely, the average probability of a bit-error at the output of the decoder is a measure of the performance of the demodulator-decoder combination. In general, the probability of error is a function of the code characters, the types of waveforms used to transmit the information over the channel, the transmitter power, and the characteristics of the channel.

As a final step, when an analog output is desired, the source decoder accepts the output sequence from the channel decoder, and, from knowledge of the source encoding method used, attempts to reconstruct the original signal from the source. Due to channel decoding errors and possible distortion introduced by the source encoder, and, perhaps, the source decoder, the signal at the output of the source decoder is an approximation to the original source output. The difference between the original signal and the reconstructed signal is a measure of the distortion introduced by the digital communication system.

2.1.3 Linear Modulation Techniques — an Overview

Digital modulation techniques may be broadly classified as linear and nonlinear. In ***linear modulation*** techniques, the amplitude of the transmitted signal, $s(t)$, varies linearly with the modulating digital signal, $m(t)$. Linear modulation techniques are bandwidth efficient and hence are very attractive for use in wireless communication systems where there is an increasing demand to accommodate more and more users within a limited spectrum.

In a linear modulation scheme, the transmitted signal $s(t)$ can be expressed as Equation (2-1), where A is the amplitude, f_c is the carrier frequency, and $m(t) = m_R(t) + \mathrm{j}\,m_I(t)$ is a complex envelope representation of the modulated signal which is in general complex form.

$$\begin{aligned} s(t) &= \mathrm{Re}[Am(t)\exp(\mathrm{j}2\pi f_c t)] \\ &= A[m_R(t)\cos(2\pi f_c t) - m_I(t)\sin(2\pi f_c t)] \end{aligned} \tag{2-1}$$

From Equation (2-1), it is clear that the amplitude of the carrier varies linearly with the modulating signal. Linear modulation schemes, in general, do not have a constant envelope. As shown subsequently, some nonlinear modulations may have either linear or constant carrier envelopes, depending on whether or not the ***base-band waveform*** is pulse shaped.

While linear modulation schemes have very good spectral efficiency, they must be transmitted using linear RF amplifiers which have poor power efficiency. Using power efficient nonlinear amplifiers leads to the regeneration of filtered sidelobes which can cause severe ***adjacent channel interference***, and results in the loss of all the spectral efficiency gained by linear modulation. However, clever ways have been developed to get around these difficulties. The most popular linear modulation techniques include pulse-shaped QPSK, OQPSK, and $\pi/4$ QPSK, which are discussed subsequently.

2.1.4 Non-Linear Modulation

Non-Linear modulation is also called ***Angle modulation.*** In Angle modulation, the angle of the carrier is varied according to the amplitude of the modulating ***baseband signal***. In this method, the amplitude of the carrier wave is kept constant (this is why FM is called constant envelope). There are a number of ways in which the phase $\theta(t)$ of a carrier signal may be varied in accordance with the baseband signal; the two most important classes of angle modulation being ***frequency modulation*** (***FM***) and ***phase modulation*** (***PM***).

Frequency modulation is a form of angle modulation in which the instantaneous frequency of the carrier signal is varied linearly with the baseband message signal $m(t)$, as shown in Equation (2-2), where A_c is the amplitude of the carrier, f_c is the carrier frequency, and k_f is the frequency deviation constant (measured in units of Hz/V).

$$s_{\mathrm{FM}}(t) = A_{\mathrm{c}} \cos[2\pi f_{\mathrm{c}} t + \theta(t)] = A_{\mathrm{c}} \cos\left[2\pi f_{\mathrm{c}} t + 2\pi k_{\mathrm{f}} \int_{-\infty}^{t} m(\eta)\mathrm{d}\eta\right] \quad (2\text{-}2)$$

If the modulation signal is a sinusoid of amplitude A_m, and frequency f_{m}, then the FM signal may be expressed as Equation (2-3).

$$s_{\mathrm{FM}}(t) = A_{\mathrm{c}} \cos\left[2\pi f_{\mathrm{c}} t + \frac{k_{\mathrm{f}} A_{\mathrm{m}}}{f_{\mathrm{m}}} \sin(2\pi f_{\mathrm{m}} t)\right] \quad (2\text{-}3)$$

Phase modulation (PM) is a form of angle modulation in which the angle $\theta(t)$ of the carrier signal is varied linearly with the baseband message signal $m(t)$, as shown in Equation (2-4).

$$s_{\mathrm{PM}}(t) = A_{\mathrm{c}} \cos[2\pi f_{\mathrm{c}} t + k_{\theta} m(t)] \quad (2\text{-}4)$$

2.1.5 FM Modulation Methods

There are basically two methods of generating an FM signal: direct method and indirect method. In the direct method, the carrier frequency is directly varied in accordance with the input modulating signal. In the indirect method, a narrowband FM signal is generated using balanced modulator; the frequency multiplication is used to increase both the frequency deviation and the carrier frequency to the required level.

1. Direct Method

In this method, ***voltage-controlled oscillators (VCOs)*** are used to vary the frequency of the carrier signal in accordance with the baseband signal amplitude variations. These oscillators use devices with reactance that can be varied by the application of a voltage, where the reactance caused the instantaneous frequency of the VCO to change proportionally.

The most commonly used variable reactance device is the ***voltage-variable capacitor*** called a varactor. The voltage-variable capacitor may be obtained, for example, by using a reverse biased ***p-n junction diode***. The larger the reverse voltage applied to such a diode, the smaller the transition capacitance will be of the diode. By incorporating such a device into a standard Hartley or Colpitis oscillator, FM signals can be generated.

2. Indirect Method

The indirect method of generating FM was first proposed by its inventor, Major Edwin Armstrong, in 1936. It is based approximating a narrowband FM signal as the sum of a carrier signal and a ***single sideband*** **(*SSB*)** signal where the sideband is 90° out of phase with the carrier. Using a ***Taylor series*** for small values of $\theta(t)$, Equation (2-5) can be expressed as

$$s_{\mathrm{FM}}(t) \approx A_{\mathrm{c}} \cos 2\pi f_{\mathrm{c}} t - A_{\mathrm{c}} \theta(t) \sin 2\pi f_{\mathrm{c}} t \tag{2-5}$$

A narrowband FM signal is generated using a balanced modulator which modulates a crystal controlled oscillator. Figure 2-3 is a direct implementation of Equation (2-5). The maximum frequency deviation is kept constant and small in order to maintain the validity of Equation (2-5), and hence the output is narrowband FM signal. A wideband FM signal is then produced by multiplying in frequency the narrowband FM signal using frequency multiplier. A disadvantage of using the indirect method for wideband FM generation is that the phase noise in the system increases with the frequency multiplying factor *N*.

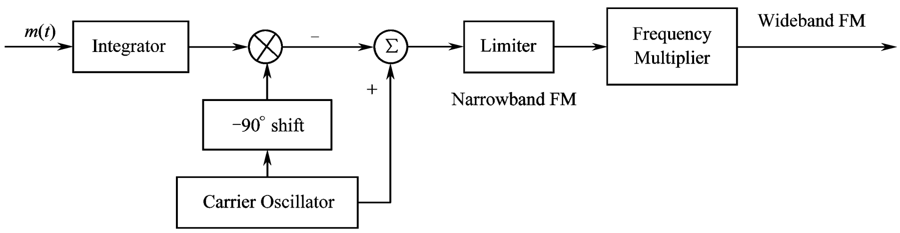


Figure 2-3 Indirect method for generating a wideband FM signal

2.1.6 FM Detection Techniques

There are many ways to recover the original information from an FM signal. The objective of all FM demodulator is to produce a transfer characteristic that is the inverse of that of the frequency modulator. That is, a frequency demodulator should produce an output voltage with instantaneous amplitude that is directly proportional to the instantaneous frequency of the input FM signal. Thus, a ***frequency-to-amplitude converter*** circuit is a frequency demodulator. Various techniques such as slope detection, zero-crossing, phase locked discrimination, and quadrature detection are used to demodulate FM. Devices which perform FM demodulation are often called frequency discriminators. In practical receivers, the RF signal is received, amplified, and filtered at the carrier, and then converted to an ***intermediate frequency*** **(*IF*)** which contains the same spectrum as the original received signal.

1. Slope Detector

It can be easily shown that FM demodulation can be performed by taking the time derivative (often called ***slope detection***) of the FM signal, followed by envelope detection. A block diagram of such an FM demodulator is shown in Figure 2-4.

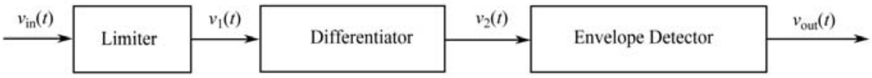


Figure 2-4 Block diagram of a slope detector type FM demodulator

The FM signal is first passed through an ***amplitude limiter*** which removes any amplitude perturbations which the signal might have undergone due to fading in the channel, and produces a constant envelope signal. Using Equation (2-6), the signal at the output of the limiter can be represented as

$$v_1(t) = V_1 \cos[2\pi f_c t + \theta(t)] = V_1 \cos\left[2\pi f_c t + 2\pi k_f \int_{-\infty}^{t} m(\eta)\,\mathrm{d}\eta\right] \tag{2-6}$$

Equation (2-6) can be differentiated in practice by passing the signal through a filter with a transfer function that has gain that increases linearly with frequency. Such a filter is called a slope filter (which is where the term slope detector derives its name). The output of the differentiator then becomes as Equation (2-7), and the output of the envelope detector becomes Equation (2-8).

$$v_2(t) = -V_1\left[2\pi f_c t + \frac{\mathrm{d}\theta}{\mathrm{d}t}\right]\sin[2\pi f_c t + \theta(t)] \tag{2-7}$$

$$v_{out}(t) = V_1\left[2\pi f_c + \frac{\mathrm{d}}{\mathrm{d}t}\theta(t)\right] = V_1 2\pi f_c + V_1 2\pi k_f m(t) \tag{2-8}$$

The Equation (2-8) shows that the output of the envelope detector contains a DC term proportional to the carrier frequency and a time-varying term proportional to the original message signal $m(t)$ .The DC term can be filtered out using a capacitor to obtain the desired demodulated signal.

2. Zero-Crossing Detector

When linearity is required over a broad range of frequencies, such as for data communications, a ***zero-crossing detector*** is used to perform frequency-to-amplitude conversion by directly counting the number of zero crossings in the input FM signal. The rationale behind this technique is to use the output of the zero-crossing detector to generate a pulse train with an average value that is proportional to the frequency of the input signal. This demodulator is sometimes referred to as pulse-averaging discriminator. A block diagram of a pulse-averaging discriminator is shown in Figure 2-5.

The input FM signal is first passed through a limiter circuit which converts the input signal to a frequency modulated pulse train. This pulse train $v_1(t)$ is then passed through a differentiator whose output is used to trigger a monostable multivibrator (also called a "one-shot"). The output of the one-shot consists of a train of pulses with average duration proportional to the desired message signal. A low pass filter is used to perform the averaging operation by extracting the slowly varying dc component of the signal at the output of the one-shot. The output of the low pass filter is the desired demodulated signal.

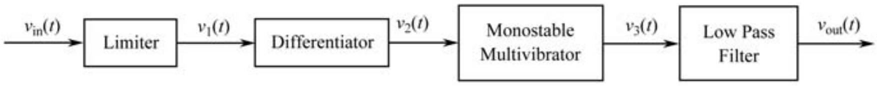


Figure 2-5 Block diagram of a zero-crossing detector

3. PLL for FM Detection

The phase locked loop (PLL) method is another popular technique to demodulate an FM signal. The PLL is a closed loop control system which can track the variations in the received signal phase and frequency. A block diagram of a PLL circuit is shown in Figure 2-6. It consists of a voltage controlled oscillator $H(s)$ with an output frequency which is varied in accordance with the demodulated output voltage level.

The output of the voltage controlled oscillator is compared with the input signal using a ***phase comparator***, which produces an output voltage proportional to the phase difference. The phase difference signal is then fed back to the VCO to control the output frequency. The ***feedback loop*** functions in a manner that facilitates locking of the VCO frequency to the input frequency. Once the VCO frequency locked to the input frequency, the VCO continues to track the variations in the input frequency. Once this tracking is achieved, the control voltage to the VCO is simply the demodulated FM signal.

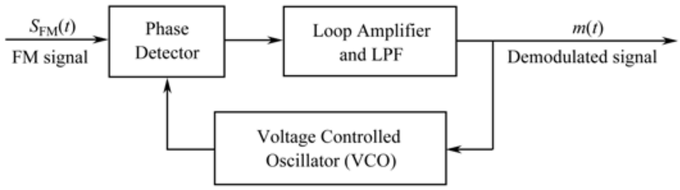


Figure 2-6 Block diagram of a PLL used as a frequency demodulator

2.1.7 Frequency Modulation & Amplitude Modulation

Frequency modulation (FM) is the most popular analog modulation technique used in mobile radio systems. In FM, the amplitude of the modulated carrier signal is kept constant while its frequency is varied by the modulating message signal. Thus, FM signals have all their information in the phase or frequency of the carrier. In amplitude modulation (AM) schemes, there is a linear relationship between the quality of the received signal and the power of the received since AM signals superimpose the exact relative amplitudes of the modulating signal onto the carrier.

Since signals are represented as frequency variations rather than amplitude variations, FM signals are less susceptible to atmospheric and impulse noise, which tend to cause rapid fluctuations in the amplitude of the received radio signal.

Also, message amplitude variations do not carry information in FM, so burst noise does not affect FM system performance as much as AM systems, provided that the FM received signal is above the ***FM threshold***. Unlike AM, in an FM system, the modulation index, and hence bandwidth occupancy, can be varied to obtain greater signal-to-noise performance. It can be shown that, under certain conditions, the FM ***signal-to-noise ratio*** improves 6dB for each doubling of bandwidth occupancy.

This ability of an FM system to trade bandwidth for SNR is perhaps the most important reason for its superiority over AM. However, AM signals are able to occupy less bandwidth as compared to FM signals.

An FM signal is a constant envelope signal, due to the fact that the envelope of the carrier does

not change with changes in the modulating signal. Hence the transmitted power of an FM signal is constant regardless of the amplitude of the message signal. The constant envelope of the transmitted signal allows efficient Class C power amplifiers to be used for RF power amplification of FM. In AM, however, it is critical to maintain linearity between the applied message and the amplitude of the transmitted signal, thus linear Class A or ***AB amplifiers***, which are not as power efficient, must be used.

While FM systems have many advantages over AM systems, they also have certain disadvantages. FM systems require a wider frequency band in the transmitting media (generally several times as large as that needed for AM) in order to obtain the advantages of reduced noise and capture effect. FM transmitter and receiver equipment is also more complex than that used by amplitude modulation systems. Both AM and FM may be demodulated using inexpensive noncoherent detectors. AM is easily demodulated using an envelope detectors whereas FM is demodulated using a discriminator or slope detector. AM may be detected coherently with a product detector, and in such cases AM can outperform FM in weak signal conditions since FM must be received above threshold.

Technical words and phrases

conventional [kən'venʃənl] *adj.* 常规的，传统的
period ['piəriəd] *n.* 时期，周期
anti-interference 抗干扰
amplitude ['æmplitju:d] *n.* 振幅
discrete [dis'kri:t] *adj.* 离散的
sample ['sæmpl] *v.* 采样
Analog / Digital conversion 模数转换
Digital/Analog conversion 数模转换
character ['kæriktə] *n.* 字符
source encoding 信源编码
channel encoding 信道编码
code rate 编码速率
demodulator [di:'mɔdjuleitə] *n.* 解调器
redundancy [ri'dʌndənsi] *n.* 冗余
distortion [dis'tɔ:ʃən] *n.* 失真
sidelobe 旁瓣
angle ['æŋgl] *n.* 角度
sinusoidal [sainə'sɔidl] *adj.* 正弦的，正弦曲线
carrier ['kæriə] *n.* 载波
baseband ['beisbænd] *n.* 基带
FM abbr. frequency modulation 调频
envelope ['envələup] *n.* 包络

phase [feiz] *n.* 相位
linearly ['liniəli] *adv.* (成)直线地，线性地
deviation [di:vi'eiʃən] *n.* 偏离，偏差；误差
phase modulation (PM) 调相，相位调制
differentiate [difə'renʃieit] *vt.* 区分；区别，辨别
index ['indeks] *n.* 指数；系数
transmitter [træns'mitə] *n.* 发射机
narrowband 窄带
oscillator ['ɔsileitə] *n.* 振荡器
wideband 宽带
phase locked loop (PLL) 锁相环
single sideband (SSB) 单边带
Taylor series 泰勒级数
demodulator [di:'mɔdjuleitə] *n.* 解调器
discrimination [diskrimi'neiʃən] *n.* 辨别；辨别力，识别力
intermediate frequency (IF) 中频
spectrum ['spektrəm] *n.* 频谱
limiter ['limitə] *n.* 限制器，限动器，限幅器
perturbations [pə:tə:'beiʃən] *n.* 扰动；微扰
slope filter 斜率滤波器
time-varying 时变的
zero-crossing detector 过零鉴别器
pulse-averaging discriminator 脉冲平均鉴别器
multivibrator [mʌlti'vaibreitə] *n.* 多频振荡器
phase comparator 相位比较器
feedback loop 反馈回路，反馈环
atmospheric ['ætməs'ferik] *adj.* 大气的；大气层的
fluctuation [flʌktʃu'eiʃən] *n.* 波动，起伏
threshold ['θreʃəuld] *n.* 门限
signal-to-noise ratio SNR 信噪比
superiority [səpiəri'ɑ:iti] *n.* 优越（性），优等

2.2 Reading Materials

1．Taiwan and China handset makers compete for mobile TV users

Taiwan- and China-based ODM handset makers are set to compete head-on for a greater share of mobile TV phone orders globally amid the emerging market for digital video broadcasting services, according to industry sources.

China-based ZTE Communications announced recently that it has signed an agreement with BT Movio (a subsidiary of BT) and Virgin Mobile under which ZTE will supply the UK-based service companies with 3G/IP-DAB (digital audio broadcasting) dual mode handsets.

ZTE's announcement puts it in competition with Taiwan-based High Tech Computer (HTC), which currently is manufacturing the Lobster 700 TV, an IP-DAB enabled mobile TV phone, for Virgin Mobile, the sources noted.

China-based TechFaith Wireless plans to showcase its first T-DMB (terrestrial-digital mobile broadcasting) enabled phone at the forthcoming 2006 China International DAB/DMB Forum and Expo, which will be held in Beijing from October 25～27, according to the company.

TechFaith is expected to make inroads in the domestic and overseas mobile TV phone market since its T-DMB phone, which will be built using mobile digital TV (MDTV) receiver solutions from Israel-based Siano Mobile Silicon, can support T-DMB, IP-DAB and DVB-H (digital video broadcasting-handheld) standards, the sources noted.

However, two Taiwan-based ODM handset makers, Compal Communications and Inventec Appliances, are also developing mobile TV handsets, using MDTV chip solutions from Siano, with Inventec expected to begin volume production of its first mobile TV phone by the end of this year, the sources stated.

(Source：DigiTimes.com)

2．World Summit on the Information Society Hailed as Resounding Success

Tunis, 18 November 2005 — The second phase of the World Summit on the Information Society closed today after almost a week of intense negotiations, eight plenary sessions, 308 parallel events organized by 264 organizations and 33 press conferences attracting around 19,000 participants worldwide.

Hailed as a resounding success by national delegations from 174 States and participants from more than 800 entities including UN agencies, private sector companies and civil society organizations, the Summit was convened in Tunis to tackle the problem of the "digital divide" and harness the potential of information and communication technologies (ICTs) to drive economic and social development.

The two Summit outcome documents — the *Tunis Commitment* and the *Tunis Agenda for the Information Society* — were endorsed by world leaders at the closing plenary of the Summit on Friday evening.

3．Equalization

In practical digital communication systems that are designed to transmit at high speed through band-limited channels, the frequency response $C(f)$ of the channel is unknown with sufficient precision to design optimum filters for the modulator and demodulator. For example, in digital communication over the dial-up telephone network, the communication channel will be different every time we dial a number, because the channel route will be different. This is an example of a channel whose interface istics are unknown a priori. There are other types of channels, e.g., wireless channels such as radio channels and underwater acoustic channels, whose frequency response

characteristics are time-variant. For such channels, it is impossible to design optimum fixed demodulation filters.

4．Motorola Opens Flagship Store in India

Motorola has opened a “Flagship Store” in India’s New Delhi. The opening was further supported by the simultaneous launch of a concept MOTOSTORE in the country’s capital and adds to the over 350 Motorola branded locations spread throughout Asia Pacific, Europe, India and Latin America. It adds to the existing Motorola Flagship stores already open in Shanghai and Moscow.

“Motorola’s Flagship Stores are designed with one mission: to help people understand and realize the full potential of Motorola products. And we have seen that when they do that, they are more likely to buy our products and more likely to be happy customers”, said Motorola’s Jeremy Dale, Corporate Vice President of Global Marketing & Communications for Motorola Mobile Devices. “MOTOSTORE is truly reinventing the mobile retail culture and is the ideal vehicle for bringing the mobile experience to life for consumers in a way that is interactive, human and fun. ”

Motorola has spent the last two years deeply engaged in a global research and development effort around the new retail concept, and as a result has come up with an approach that is centered around the consumer.

“India has one of the fastest mobile adoption rates in the world and as part of its massive growth; we recognize that mobile users in India desire a wide range of devices and services. In the flagship store, consumers will be able to personalize their Motorola handsets by downloading ringtones, games, wall papers and other applications, ” said Jason Few, Corporate Vice President of Retail & Distribution for Motorola Mobile Devices.

Corporate Vice President of Global Marketing & Communications for Motorola Mobile Devices.

（Source: cellular-news）

5．Do you trust online transactions?

Geneva, 25 April 2006 — In today’s interconnected and increasingly networked world, we are vulnerable to a wide variety of threats, including deliberate attacks on critical information infrastructures, with debilitating effects on our economies and on our societies. Is fear of identity theft, the misuse of sensitive personal data and other online fraud having an impact on confidence in the use of the internet for commercial or financial purposes? Is the willingness to conduct online transactions outpacing the trust? To what extent are businesses and consumers willing to take risks in order to reap the benefits of online transactions?

ITU is conducting a worldwide public survey to assess users’ trust of online transactions and awareness of cybersecurity measures. The data collected through the survey will be used to increase global awareness of cybersecurity, particularly in developing countries, and should help decision-makers in assessing the cyberspace “trust” level with a view to reviewing national and corporate strategies and priorities.

The lack of adequate security is an obstacle for using ICTs that rely on the protection and confidentiality of sensitive data. These security and trust issues are already beginning to have an adverse impact on the social and economic development of the world.

2.3 Exercises

1. Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）模拟通信系统
（2）数字通信系统
（3）模/数转换
（4）基带波形
（5）编码效率
（6）邻道（频）干扰
（7）基带信号
（8）相位调制
（9）压控振荡器
（10）AB 放大器
（11）PN 结二极管
（12）单边带信号
（13）信噪比
（14）泰勒级数
（15）频幅转换器
（16）中频
（17）过零鉴别器
（18）比相器
（19）反馈环
（20）调频门限

2. Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）According to the type of transmitted signal, modern communication can be divided into（　　）communication and（　　）communication.
（2）Digital signal is a kind of signal, which has limited number for its （　　）.
（3）The Analog/Digital conversion contains the following three steps:（　　）,（　　） and（　　）.
（4）The process of converting the output of either an analog or digital source into a sequence of binary digits is called（　　）or （　　）.
（5）The channel encoder introduces some（　　）in the binary information sequence, which can be used at the receiver to overcome the effects of（　　）and （　　）.

(6) Digital modulation techniques may be broadly classified as (　　) and (　　).

(7) A percentage of modulation greater than 100% will (　　) the message signal if detected by an (　　) detector.

(8) While linear modulation schemes have very good spectral (　　), they must be transmitted using linear RF amplifiers which have (　　) power efficiency, leading to the regeneration of filtered sidelobes which can cause severe (　　), and results in the (　　) of all the spectral efficiency.

(9) In Angle modulation, the angle of the carrier is varied according to the (　　) of the modulating (　　).

(10) The two most important classes of angle modulation being (　　) and (　　).

(11) The two methods to generate an FM signal are (　　) method and (　　) method.

(12) Various techniques such as (　　), (　　), (　　), and (　　) are used to demodulate FM.

(13) An FM signal is a constant (　　) signal, and the transmitted power of an FM signal is constant regardless of the (　　) of the message signal.

(14) FM systems require a wider (　　) in the transmitting media in order to obtain the advantages of reduced noise and capture effect.

3. Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

(1) (　　) If the signals are continuous for amplitude, they are called digital signals.

(2) (　　) The efficiency of channel bandwidth usage for analog communication system is higher than that of the digital communication system.

(3) (　　) It is not necessary for the analog signals be converted into digital signals before they are processed and transmitted in digital communication systems.

(4) (　　) The receiving signal in the receiving end of a digital communication system must have to be converted into analog signals by a Digital/Analog conversion.

(5) (　　) Linear modulation techniques are bandwidth efficient.

(6) (　　) Nonlinear modulation are very attractive for use in wireless communication systems where there is an increasing demand to accommodate more and more users within a limited spectrum.

(7) (　　) Frequency modulation is a form of linear modulation.

(8) (　　) The most commonly used variable reactance device is the voltage-variable capacitor.

(9) (　　) The indirect method of generating FM was first proposed in 1946.

(10) (　　) A zero-crossing detector is used to perform frequency-to-amplitude conversion when linearity is required over a broad range of frequencies.

(11) (　　) FM signals have all their information in the phase or frequency of the carrier.

(12) (　　) AM signals are less susceptible to atmospheric and impulse noise compared to FM signals.

（13）（　　）In an FM system, the bandwidth occupancy can be varied to obtain greater signal-to-noise performance.

（14）（　　） AM signals are able to occupy less bandwidth as compared to FM signals.

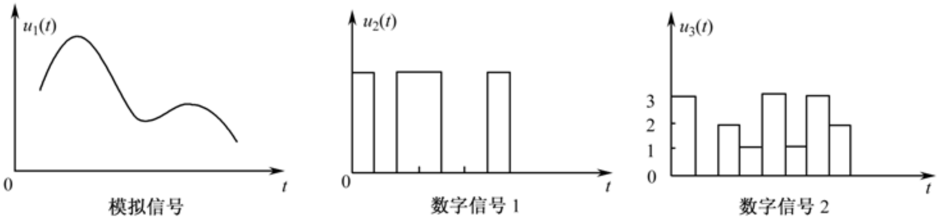
2.4　课文参考译文　线性调制技术

2.4.1　数字通信与模拟通信

根据传输信号的不同，通信可分为模拟通信和数字通信。当一个通信系统传输处理的信号是模拟信号时，这个系统就是模拟通信系统。反之，数字通信系统就是传输、处理数字信号的系统。

传统的电话通信系统中，用户线上传送的信号在整个通话期间不断地随着用户声音的变化而变化。这个变化的信号无论在时间上或是在幅度上都是连续的，这种信号称为模拟信号，如图 2-1（a）所示。数字信号与模拟信号不同，它是一种时间上离散的、幅度取值有限的信号形式。常见的数字信号是幅度取值只有两种（用 0 和 1 代表）的波形，称为“二进制信号”，如图 2-1（b）所示。一般地，数字信号可以用二进制、M 进制（$M>2$）来表示，显然，M 进制信号就是有 M 种幅度符号可选的数字信号。图 2-1（c）所示就是 4 进制信号，有 0、1、2、3 共 4 个可选幅度。

由于模拟信号的频谱较窄，模拟通信系统的信道利用率较高。但因为连续信号中混入噪声后很难清除，使得输出的还原信号产生波形失真，模拟通信系统的抗干扰能力只能算是差强人意。



译图 2-1　模拟信号与数字信号

现代通信系统大多是数字通信系统，因此，模拟信号必须首先变为数字信号后才能进入数字通信系统传输，这个模拟转换为数字信号的过程就称为“模/数转换”。它一般包括三个过程:

① 抽样：就是对连续的模拟信号进行离散化处理；

② 量化：将模拟信号抽样值转换为离散值；

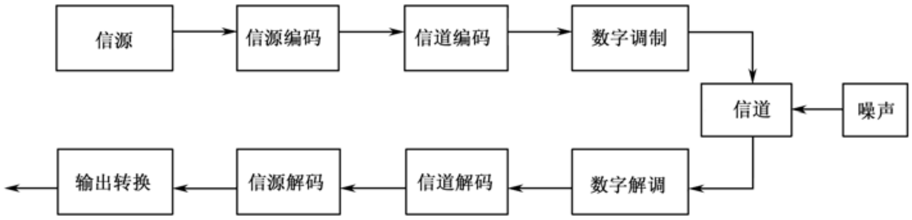
③ 编码：把量化后的离散信号用一组二进制代码表示，形成数字信号。

接收端对收到信号的处理则正好是上述过程的逆过程，也就是把收到的数字信号还原为原始模拟信号，即“数模变换”。

2.4.2　数字通信系统的基本组成

图 2-2 给出了数字通信系统得基本框图及其组成。图中，信源既可以输出模拟信号（如音频信号），也可以输出数字信号（如打字机的输出信号），这是一类时间上离散且总的个数

有限的信号。数字通信系统中，信源输出消息（无论模拟还是数字的）都将被转换为二进制数字序列，这一转换过程就叫做信源编码或数据压缩，通常希望表示信源输出消息的符号个数越少越好，换句话说，信源编码应用较少的符号有效（或无冗余地）表示信源输出消息。



译图 2-2　数字通信系统的组成框图

信源编码后的二进制数字序列，常称之信息序列，将被送入信道编码器进行信道编码。这是因为信号在信道传输过程中将受到各种噪声和干扰的影响，信道编码在传输的二进制信息序列中增加一些冗余，以便接收端利用这些冗余克服噪声干扰的不利影响，正确恢复原始信号。所以，信道编码中增加的冗余可提高接收信号的可靠性和正确性，帮助接收机解调还原所传的信息序列。信道编码一次性地将 k 比特信息序列按一定的规则变换为一个 n 比特的信息序列，这个变换后的信息序列称为码组。衡量信道编码所带来的冗余度可以用 n/k 这个比值，这个比值的倒数 k/n 就称为该信道编码的编码效率。

信道编码输出的二进制序列经过数字调制器调制后被送入信道进行传输。鉴于目前所用的信道绝大部分都是传输电信号的，可以说数字调制器的作用主要就是将二进制信息序列转换为适合信道传输的信号波形。

数字通信系统的接收端，解调器将收到的变形、失真波形解调还原为二进制或 M 进制符号序列后，再送入信道解码器，借助其中的信息冗余，按照事先所知的信道编码规则重构原始信息序列。

常用接收端恢复的信息序列中的错误发生概率来衡量解调与解码的性能，具体地，就是解码器输出序列中平均发生一个比特误码的概率——误码率。实际上，误码率不但与接收端解调器、解码器的性能有关，也同样与发送端所选编码的特性、传输波形的类型、发送信号的功率以及信道的性能等有关。

最后，若输出端要求输出模拟信号，信道解码器输出序列送入信源解码器，利用已知的信源编码方式还原输出原始信号。信道解码错误以及信源编码/解码差错都将导致信源解码输出仅仅是发送端信源输出原始消息的一个近似，一般常用这两者之间的差异表示数字通信系统的失真。

2.4.3　线性调制技术一览

数字调制可大致分为线性调制和非线性调制，线性调制信号 $s(t)$ 的幅度随调制信号 $m(t)$ 的幅度发生线性变化。线性调制技术的频带利用率较高，能满足越来越多的用户不断增长的频带需求，因而在无线通信中得到了广泛的应用。

线性调制的信号 $s(t)$ 可表示为式（2-6）所示形式。其中，A 为幅度，f_c 为载波频率，$m(t)$ 则是已调信号包络的通用复数表示形式，即 $m(t)=m_R(t)+\mathrm{j}m_I(t)$。

$$\begin{aligned}s(t)&=\mathrm{Re}[Am(t)\exp(\mathrm{j}2\pi f_c t)]\\&=A[m_R(t)\cos(2\pi f_c t)-m_I(t)\sin(2\pi f_c t)]\end{aligned}\tag{2-1}$$

由式（2-1）可知，线性调制中，载波信号的幅度随调制信号的变化而线性变化。

尽管线性调制的频谱效率较高，但其信号发射却必须使用低效率的线性射频放大器，因为若采用高效率的非线性放大器，将使被滤除的旁瓣信号分量再生，导致严重的邻道干扰，进而使线性调制的整个频带效率降低。对此问题，目前已有不少解决办法，包括常用的线性调制技术，如四相相移键控、交错正交相移键控、$\pi/4$ 正交相移键控等。

2.4.4　非线性调制

非线性调制也称角调制，其载波信号的相角随着基带调制信号的幅度变化而变化的调制方式，而载波的幅度保持不变（故常称 FM 调制为恒包络调制）。实现载波信号的相角随基带信号而变的方法有多种，其中两个最重要的就是频率调制和相位调制。

频率调制（FM）是角调制的一种特殊形式，其载波信号的瞬时角频率随线性基带调制信号 $m(t)$变化的关系如式（2-2）所示。其中，A_c 为载波的幅度；f_c 为载波频率；k_f 为频率偏移常数（单位是 Hz/V）。

$$s_{FM}(t) = A_c \cos[2\pi f_c t + \theta(t)] = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_{-\infty}^{t} m(\eta) d\eta\right] \quad （2-2）$$

若调制信号为幅度 A_m、频率 f_m 的正弦信号，FM 信号可表示为式（2-3）。

$$s_{FM}(t) = A_c \cos\left[2\pi f_c t + \frac{k_f A_m}{f_m}\sin(2\pi f_m t)\right] \quad （2-3）$$

相位调制是使载波信号的相位 $\theta(t)$ 随基带调制信号 $m(t)$ 的变化而线性变化的角调制，其关系如式（2-4）所示。

$$s_{PM}(t)=A_c\cos[2\pi f_c t+k_\theta m(t)] \quad （2-4）$$

2.4.5　频率调制

产生调频信号有两种基本方法：直接调频法和间接调频法。前一方法使载波的频率直接随着输入调制信号的变化而变化；间接调频法首先通过平衡调制器产生窄带 FM 信号，再利用倍频器将频偏及载频提升到所需数值。

1. 直接调频法

直接调频法利用压控振荡器（VCO）使载波频率随基带信号的幅度变化而变。这些振荡器包含了随电压变化而变的电抗器件，从而使得振荡器的瞬时频率与输入信号的电平成比例地变化。

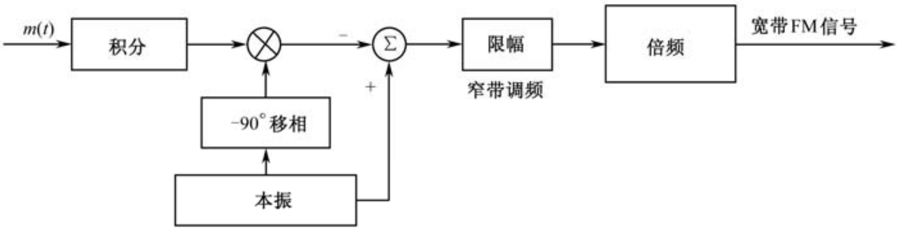
最常用的电抗器件是压变电容，常称为变容二极管，可由反偏的 PN 结二极管构成，当反偏电压越大时，二极管的传导容量就越小。将此反偏二极管装入标准的哈特莱振荡器或考毕兹振荡器，即可获得调频信号。

2. 间接调频法

1936 年，埃德温•阿姆斯特朗首次提出了间接调频方式，利用一个载波信号和单边带（SSB）信号叠加来近似窄带调频信号。其中，单边带信号的相位超前于载波 90º。当相角 $\theta(t)$ 的值较小时，利用泰勒级数可写出式（2-5）的调频信号。

$$s_{PM}(t)\approx A_c\cos[2\pi f_c t-k_c\,\theta(t)\sin2\pi f_c t \quad （2-5）$$

间接调频法通过平衡调制器产生窄带调频信号，其基本框图如图 2-3 所示，其平衡调制器由晶体控制振荡衡实现。该框图完全基于式（2-5）得出，为满足式（2-5）的前提条件，其最大频偏较小且保持不变，故其输出信号为窄带 FM 信号。利用倍频器，可将该窄带 FM 信号生成宽带 FM 信号。对宽带调频而言，该间接调制方法的缺点是系统的相位噪声随倍频倍数 N 的增加而增大。



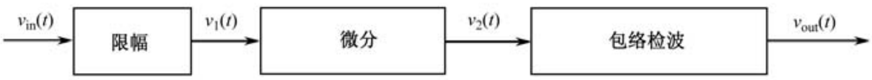
译图 2-3　间接宽带调频器

2.4.6　频率解调

调频信号的解调方法较多，究其目的，都是通过构造解调器的传输特性，使其实现频率调制的逆过程。也就是说，频率解调器输出的瞬时信号幅度电平应与输入解调信号的瞬时频率成正比，故频率解调器就是一个频率–幅度转换电器。有多种技术都可用于调频信号解调，如斜率跟踪、过零检测、锁相鉴频、正交鉴频等，其实现电路就称为鉴频器。在实际接收器中，射频信号接收后，通过放大，并滤除载波分量，然后转换为一个中频信号，其包含了与原调制信号相同的频谱，就可恢复原始信号。

1. 斜率鉴频法

对调频信号求导（常称之为斜率检波）后，再对其进行包络检波，可以实现频率解调，如图 2-4 所示。



译图 2-4　斜率鉴频法框图

图中，调频信号首先通过限幅器，去除所有可能因信道衰落引起的振幅扰动信号后，输出一个恒包络信号，如式（2-6）所示。

$$v_1(t) = V_1\cos[2\pi f_c t + \theta(t)] = V_1\cos\left[2\pi f_c t + 2\pi k_f\int_{-\infty}^{t} m(\eta)\mathrm{d}\eta\right] \tag{2-6}$$

对式（2-6）中的信号进行微分，实际中通过将信号送入滤波器获得其频率增益来实现。实现该功能的滤波器就是斜率滤波器（由斜率检波器而得名），微分器输出信号如式（2-7），所示，包络检波输出信号如式（2-8）所示。

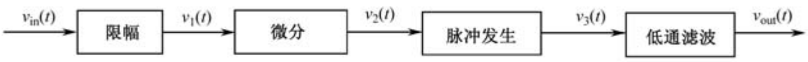
$$v_2(t) = -V_1\left[2\pi f_c t + \frac{\mathrm{d}\theta}{\mathrm{d}t}\right]\sin[2\pi f_c t + \theta(t)] \tag{2-7}$$

$$v_{out}(t) = V_1\left[2\pi f_c + \frac{\mathrm{d}}{\mathrm{d}t}\theta(t)\right] = V_1 2\pi f_c + V_1 2\pi k_f m(t) \tag{2-8}$$

由式（2-8）可以看出，包络检波输出信号包含一个与载波频率成正比的直流分量，以及一个与原始信号 $m(t)$ 成正比的时变分量。通过电容滤掉该直流分量后，就可获取所需的解调信号。

2. 过零鉴频法

当数据通信中对（信号解调的）线性要求高于带宽要求时，可采用过零鉴频法，通过直接对输入 FM 信号的过零点计数，来实现频率—振幅转换。该方法的基本思路是利用过零鉴频的输出（过零点数），生成幅度均值与输入 FM 信号频率成正比的脉冲序列。过零鉴频器有时也称为脉冲平均鉴别器，如图 2-5 所示。

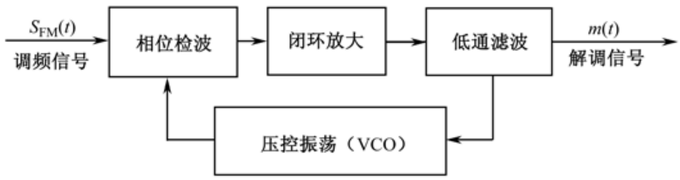


译图 2-5　过零鉴频器框图

图中，输入 FM 信号首先通过限幅器限幅，成为频率受调的脉冲序列；该脉冲调制序列 $v_1(t)$ 再通过微分后送入单稳触发器（也称为“one-shot”），触发该“one-shot”输出周期正比于调制信号的脉冲序列；再经过低通滤波器，提出“one-shot”输出信号中缓慢变化的直流分量，即得到所需 FM 解调信号。

3. 锁相环鉴频法

利用锁相环（PLL）进行 FM 解调也是一个常用方法。锁相环是一个闭环控制系统，可以跟踪（解调器）接收信号的相位和频率变化。锁相环鉴频器框图如图 2-6 所示，它含有一个（传输函数为）$H(s)$ 的压控振荡器（VCO），其输出信号的频率随解调输出信号的电压变化而变化。



译图 2-6　锁相环解调频框图

压控振荡器（VCO）的输出是输入 FM 信号经过比相而来，其电平值正比于信号的相位差。该相差信号被馈送回 VCO，控制 VCO 输出信号的频率。通过反馈环路，VCO 的振荡频率逐渐接近输入信号频率，当 VCO 的振荡频率等于输入信号频率后，它就能持续跟踪输入频率的变化。此时，压控振荡器（VCO）的控制电压信号就是 FM 解调输出信号。

2.4.7　调频与调幅

调频是移动无线通信系统中使用最广的模拟调制技术，其已调信号的幅度保持恒定而频率则随调制信号的变化而变，故其（有效）信息全部包含在载波的频率或相位中。调幅（AM）方式由于将调制信号完全叠加在载波信号上，其接收信号的质量与其功率成正比关系。

由于调频信号的振幅不变而是频率随调制信号而变，调频信号不易受到大气噪声和脉冲噪声的影响，这两类噪声容易引起接收无线信号的幅度发生快速波动。

由于调频信号的振幅并不携带有效信息，只要收到的 FM 信号电平高于其阈值，突发噪声对调频系统性能的影响远小于 AM 系统。这是因为与 AM 系统不同，FM 系统可以通过改变调频指数、带宽来获得更大的信噪比。实践表明，在某些情况下 FM 系统带宽每增加 1 倍，其输出信噪比将提高 6dB。

以带宽换取信噪比的能力可能是 FM 系统相比于 AM 系统的最大优势，但 AM 信号占用的带宽小于 FM 信号。

由于载波信号的包络不随调制信号的变化而变化，调频信号是恒包络信号，不论调制信号的幅度多大，调频信号的发射功率总是恒定不变的。由于 FM 信号包络恒定，可选择效率较高的 C 类功放进行射频功率放大。而 AM 系统由于必须保持发射信号与调制信号之间的幅度线性关系，而必须选用效率较低的 A 类或 AB 类放大器。

尽管 FM 系统比 AM 系统具有很多优势，但也有一些不足。调频系统需要带宽更宽（通常是 AM 系统的几倍），才能获得其噪声低、易于捕捉的优势。调频系统的发射和接收设备也比调幅系统复杂得多。虽然 AM、FM 系统都可以采用廉价的非相干解调器方式，但 AM 信号只需通过包络检波即可很容易地实现解调，而 FM 信号则需要通过斜率跟踪或鉴频器才能解调。由于调频信号必须在输入信号高于其阈值电平的情况下才能解，而 AM 信号则无此门限，因此弱信号条件下 AM 系统的性能优于 FM 系统。

Unit 3　Coding Techniques

3.1　Text

3.1.1　Source coding

Communication systems are designed to transmit the information generated by a source to some destination. Information source may take a variety of different forms. For example, in radio broadcasting, the source is generally an audio source (voice or music). In TV broadcasting, the information source is a video source whose output is a moving image. The outputs of these sources are analog signals and, hence, the sources are called analog sources. In contrast, computers produce discrete outputs (usually binary or ASCII characters) and, hence, they are called discrete sources.

Whether the source is analog or discrete, modern digital communication system is generally designed to transmit information in digital form. Consequently, the output of the source must be converted to a format that can be transmitted digitally. This conversion is generally performed by the source encoder, whose output may be assumed to be a sequence of binary digits.

The process of representing the source output is a process of representing the source output by a sequence of binary digits. When X is the output of a discrete source, the entropy $H(X)$ of the source represents the average amount of information emitted by the source. A measure of the efficiency of a ***source-encoding*** method can be obtained by comparing the average number of binary digits per output letter from the source to the entropy $H(X)$.

In any case, we shall demonstrate that it is always more efficient to encode blocks of symbol instead of encoding each symbol separately. By making the block size sufficiently large, the average number of binary digits per output letter from the source can be made arbitrarily close to the entropy of the source. The two famous sources encoding methods are ***Shannon-Fano source encoding algorithm*** and ***Huffman source encoding algorithm***. Let us illustrate these two encoding algorithms by means of the following two examples.

Exp. 3-1　Consider a ***DMS*** **(*****Discrete Memoryless Source*****)** with eight possible symbols A、B、C、D、E、F、G、H having the probabilities of occurrence as following. Now we can begin the encoding process according to *Shannon-Fano source encoding algorithm* as the following six steps.

X:	A	B	C	D	E	F	G	H
$P(X)$:	0.01	0.27	0.09	0.14	0.05	0.12	0.03	0.29

① Re-arrange the all symbols according to the falling sequence of their occurring probabilities;

② Divide the symbols into two groups, making sum of probabilities in one group equal to another as close as possible;

③ Re-divide the symbols in each group separately according to the same rule in step ②, and then we get four groups;

④ Repeat the same process as step ③ till the each symbol is separated alone;

⑤ Assign 0 to all the first group messages at each division, and 1 to the second group messages;

⑥ The array for each symbol of its all assigned 0 and 1 forms the Shannon-Fano coding we need.

The whole process of Shannon-Fano Coding method is shown in the following Table 3-1.

Table 3-1 Example for Shannon-Fano Coding Algorithm

signal	probability	First class	Second class	Third class	Forth class	Fifth class	Code group	Group length
H	0.29	0	0				00	2
B	0.27	0	1				01	2
D	0.14	1	0	0			100	3
F	0.12	1	0	1			101	3
C	0.09	1	1	0			110	3
E	0.05	1	1	1	0		1110	4
G	0.03	1	1	1	1	0	11110	5
A	0.01	1	1	1	1	1	11111	5

Exp. 3-2 Consider the same DMS with eight possible symbols in example 3-1, now we begin the encoding process according to Huffman source encoding algorithm as the following five steps.

① Re-arrange the eight symbols according to the falling sequence of their occurring probabilities;

② Tie the two least probable symbols together with the upper branch assigned a 1 and the lower assigned a 0. Then add the two probabilities together and regard these two symbols as a new symbol;

③ Re-arrange the seven symbols according to the falling sequence of their occurring probabilities;

④ Repeat the same process as step ② till the all probabilities have been processed;

⑤ Beginning with the left node to the last right node in the following Figure 3-1, record the all binary data from bottom to top, then the binary sequence written by you is what you want.

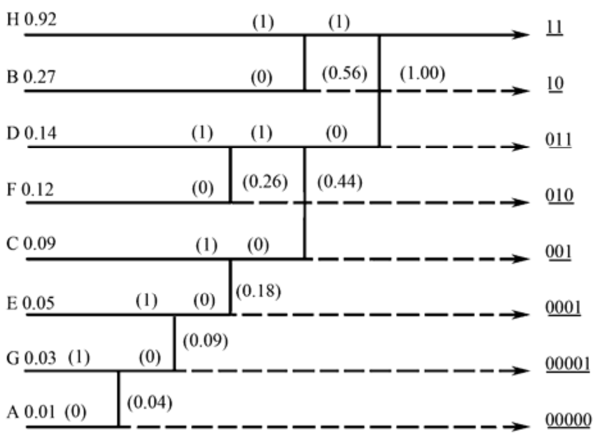


Figure 3-1 Huffman Coding process

The whole process of Huffman Coding method in exp.3-2 is as shown as the following Figure 3-1,

and the respective code words for symbols in the appointed DSM we get is listed in Table 3-2.

Table 3-2 Respective Code Words of the DSM in Exp. 3-2

symbol	A	B	C	D	E	F	G	H
Code word	00000	10	001	011	0001	010	00001	11

By applying the ***sampling theorem***, the output of an analog source is converted to an equivalent discrete-time sequence of samples. The samples are then quantized in amplitude and encoded. One type of sample encoding is to represent each discrete amplitude level by a sequence of binary digits. Hence, if we have L levels, we need $R = \log_2 L$ bits per sample if L is a power of 2, or $R = \log_2 L + 1$, if R is not a power of 2. On the other hand, if the levels are not equally probable, and the possibilities of the output levels are known, we may use Huffman coding to improve the efficiency of the encoding process.

3.1.2 Channel Capacity and Channel Coding

1. Introduction

The major significance of the channel capacity formulas is that they serve as upper limits on the transmission rate for reliable communication over a noisy channel. The fundamental rate that the channel capacity plays is given by the noisy channel coding theorem due to Shannon (1948).

There exist channel codes (and decoders) that make it possible to achieve reliable communication, with as small an error probability as desired, if the transmission rate $R < C$, where C is the ***channel capacity***. If $R > C$, it's not possible to make the probability of error tend toward zero with any code.

The design of coded modulation for efficient transmission of information may be divided into two basic approaches. One is the algebraic approach, which is primarily concerned with the design of coding and decoding techniques for specific classes of codes, such as ***Cyclic block codes*** and ***Convolutional Codes.*** The second is the ***probabilistic approach***, which is concerned with the analysis of the performance of a general class of coded signals. This approach yields bounds on the probability of error that can be attained for communication over a channel having some specified characteristic.

A block code consists of a set of fixed-length vectors called ***code words.*** The length of a word is the number of elements in the vector and is denoted by n. The elements of a code word are selected from an alphabet of q elements. When the alphabet consists of two elements, 0 and 1, the code is a binary code and the elements of any code word are called bits. When the elements of a code word are selected from an alphabet having q elements ($q > 2$), the code is non-binary.

It is interesting to note that when q is a power of 2, i.e., $q = 2^b$ where b is a positive integer, each $q-ary$ element has an equivalent binary representation consisting of b bits, and, thus, a ***non-binary code*** of block length N can be mapped into a binary code of block length $n = b \cdot N$.

There are 2^n possible code words in a binary block code of length n. From these 2^n code words, we may select $M = 2^k$ code words ($k < n$) to form a code. Thus, a block of k

information bits is mapped into a code word of length n selected from the set of $M = 2^k$ code words. We refer to the resulting block code as (n,k) code, and the ratio $\eta = k/n$ is defined to be the rate of the code. More generally, in a code having q elements, there are q^n possible code words. A subset of $M = 2^k$ code words may be selected to transmit k bit blocks of information.

Besides the code rate parameter η, an important parameter of a code word is its *weight*, which is simply the number of nonzero elements that it contains. In general, each code word has its own weight. The set of all weights in a code constitutes the *weight distribution* of the code. When all the $M = 2^k$ code words have equal weight, the code is called a ***fixed-weight code*** or a ***constant-weight code.***

2. Cyclic Codes

Cyclic Codes are a subset of the class of linear codes that satisfy the following cyclic shift property: if A=($a_{n-1}a_{n-2} \dots a_1 a_0$) is a code word of a cyclic code, then $(a_{n-2}a_{n-3} \dots a_0 a_{n-1})$, obtained by a cyclic shift of the elements of A, is also a code word. So can we know the $(a_{n-3}a_{n-4} \dots a_0 a_{n-1} a_{n-2})$, etc., are code words of this cyclic code too. In a word, all cyclic shifts of A are code words. As a consequence of the cyclic property, the codes process a considerable amount of structure which can be exploited in the encoding and decoding operations.

Since the minimum separation between a pair of code words is $d_{\min}$, it is possible for an error pattern of weight $d_{\min}$ to transform one of these 2^k code words in the code into another code. When this happens we have an ***undetected error.*** On the other hand, if the actual number of errors is less than $d_{\min}$, the syndrome will have a nonzero weight. When this occurs, we have detected the presence of one or more errors on the channel. Clearly, the (n,k) block code is capable of detecting $d_{\min}-1$ errors. Error detection may be used in conjunction with an ***automatic repeat-request* (*ARQ*)** scheme for retransmission of the code word.

The ***error correction capability*** of a code also depends on the minimum distance $d_{\min}$. However, the number of correctable error patterns is limited by the number of possible syndromes or coset leaders in the standard array. To determine the error correction capability of an (n,k) code, it is convenient to view the 2^k code words as points in an n-dimensional space. If each code word is viewed as the center of a sphere of radius (***Hamming distance***) t, the largest value that t may have without intersection (or tangency) of any pair of the 2^k spheres is $t = \frac{1}{2}\lfloor d_{\min}-1 \rfloor$, where $\lfloor d_{\min}-1 \rfloor$ denotes the largest integer contained in $(d_{\min}-1)$. Within each sphere lie all the possible received code words of distance less than or equal to t from the valid code word. This implies that an (n,k) code with minimum distance $d_{\min}$ is capable of correcting $t = \frac{1}{2}\lfloor d_{\min}-1 \rfloor$ errors.

Clearly, to correct t error implies that we have detected t errors. However, it is also possible to detect more than t errors if we compromise in the error correction capability of the code. In general, a code with minimum distance $d_{\min}$ can detect e_d errors and correct e_c errors, where

$$e_d + e_c \leqslant d_{\min} - 1 \qquad (3\text{-}1)$$

and

$$e_c \leqslant e_d \qquad (3\text{-}2)$$

Technical words and phrases

variety [və'raiəti] *n.* 变化，多样性
radio ['reidiəu] *n.* 无线电
broadcast ['brɔ:dkæst] *n.* 广播，播音；广播节目
image ['imædʒ] *n.* 图像
analog source 模拟信源
discrete source 离散信源
format ['fɔ:mæt] *n.* 形式；[计] 格式
entropy ['entrəpi] *n.* [物] 熵，[无] 平均信息量
represent [repri'zent] *vt.* 表现，描绘
average ['ævəridʒ] *adj.* 一般的，通常的，平均的
emit [i'mit] *vt.* 发出，放射，散发
demonstrate ['demənstreit] *vi.* 论证，展示，演示
block [blɔk] *n.* 块
arbitrarily *adv.* 任意地
Shannon-Fano source encoding algorithm 香农–范诺信源编码法
Huffman source encoding algorithm 霍夫曼信源编码法
DMS (Discrete Memoryless Source) 离散无记忆信源
occurrence [ə'kʌrəns] *n.* 发生，出现，事件；发生的事情
separately ['sepəritli] *adv.* 个别地；分离地
upper ['ʌpə] *adj.* 上面的
apply [ə'plai] *vt.* 应用
equivalent [i'kwivələnt] *adj.* 相等的，相当的，同意义的
power ['pauə] *n.* [数] 乘方，幂
significance [sig'nifikəns] *n.* 意义，重要性；价值
formula ['fɔ:mjulə] *n.* 公式，规则
fundamental [fʌndə'mentl] *adj.* 基础的；基本的
approach [ə'prəutʃ] *n.* 方法，步骤，途径，通路
algebraic [ældʒi'breiik] *adj.* 代数的，关于代数学的
Cyclic block codes 循环码
Convolutional Code 卷积码
probabilistic [prɔbəbi'listik] *adj.* 概率论的
bound [baund] *n.* 范围，限度
vector ['vektə] *n.* [数] 向量，矢量
subset ['sʌbset] *n.* [数] 子集
fixed-weight code (constant-weight code) 等重码（恒重码）
cyclic shift 循环移位
exploit [iks'plɔit] *vt.* 开拓，开发，开采
hard-decision 硬（件）译码

error pattern　错误图样
undetected error　未被检验出的错误
syndrome　['sindrəum]　*n.*　常见的共存情况；并存特性
conjunction　[kən'dʒʌŋkʃən]　*n.*　连接词；时机
automatic repeat-request (ARQ)　自动请求重发
error correction capability　纠错能力
array　[ə'rei]　*n.*　排列，编队
n-dimensional　*n* 维的
sphere　[sfiə]　*n.* 球；球形；领域
radius　['reidjəs]　*n.*　半径，范围
Hamming distance　汉明距
Valid　['vælid]　*adj.*　有根据的；有效的
imply　[im'plai]　*vt.*　暗示；暗指；包含
detect　[di'tekt]　*vt.*　察觉，发觉，侦查，探测

3.2 Reading Materials

1. Chinese Telecom Service Open to Foreigners

Xi Guohua, vice minister of China's Ministry of Information Industry, reiterated at the Annual Conference 2007 of the Boao Forum for Asia that China has already opened its value-added telecom service to foreign companies.

Xi said that until now, China has approved the establishment of 22,000 value-added telecom companies and value-added telecom service has become an important part of China's telecom industry. He said that breaking up monopolization and introducing competition are conducive to the development of China's telecom industry as well as the national economy, so China welcomes foreign telecom operators to come to the country under the premise of its commitments to the World Trade Organization.

2006 was when China finished its five-year transitional period for joining the WTO and by that year China had approved foreign companies to hold a maximum of 49% of the stake in the basic telecom service of Chinese companies and up to 50% of stake in the telecom value-added services.

（Source: China Tech News）

2. Motorola Hit with US$23 Million Legal Fees

A Florida judge has ruled that Motorola must pay nearly US$23 million in legal fees to lawyers who "lost" a lawsuit against the company. Judge Leroy Moe granted US$ 20 million in fees and US$ 2.9 million in costs. SPS Technologies had tried to sue Motorola for US$ 10 billion after it claimed that the company had stolen its technology and abandoned a motoring system which would have linked GPS with roadside assistance.

The Judge imposed the charges after the trial collapsed when Motorola apparently let a witness read trial transcripts, in violation of the Judge's order.

Motorola’s violation “goes right to the heart of the civil justice system and it certainly has to be addressed,” Judge Moe said last Thursday.

Celebrity lawyer, Willie Gary had asked for US$11,000 an hour, or $ 24.5 million for his work-and was seeking punitive payments which could have doubled his payment to over US$90 million. The Judge did not specify the hourly rate he was awarding, but if the award is split as expected with two-thirds going to Willie Gary, then he would have been on a rate of around US$ 6,000 per hour.

“I feel the judge sent a message to corporate America and to Motorola that you can’t cheat and get away with it,” Gary said after the ruling.

Gary has previously won over 150 multi million dollar suits, including a $ 500 million award against a Canadian Funeral Home. Other “victims” include Disney and Anheuser-Busch. He famously owns two private jets, named the Wings of Justice and the Wings of Justice II - the later of which was recently refitted with an 18-carat gold sink and $1.2 million sound system.

（Source: cellular-news）

3. ITU and European Commission to Create Investment Environment for ICT

Geneva, 17 December 2007 — The International Telecommunication Union and the European Commission (EC) concluded an agreement aimed at attracting massive investment in ICT infrastructure and ICT-enabled applications.

Over the past decade, most countries in Africa, Asia-Pacific and the Caribbean have initiated reforms in the telecommunication sector by establishing national regulatory bodies, introducing competition. However, large sections of the population remain without basic access to information and communication technology (ICT) services. Key reforms have yet to be undertaken in many countries which would provide regulators with the tools and authority to effectively regulate the sector as a means of boosting investment, promoting innovation and building confidence in ICT markets.

The European Union has allocated Euro 8 million from the European Development Fund, to which ITU will add USD 500 000 of its own resources. The work will be managed and implemented by ITU.

3.3 Exercises

1. Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）霍夫曼信源编码法
（2）模拟信源
（3）离散无记忆信源
（4）信道容量
（5）码组
（6）等重码/定比码

（7）循环码
（8）纠错能力
（9）离散信源
（10）香农–范诺信源编码法
（11）抽样编码
（12）卷积码
（13）似然法则
（14）非二进制码
（15）自动前向纠错
（16）汉明距

2. Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）When the output of a signal source is（　　）signals, the source is analog sources. In contrast, signal source producing discrete outputs, usually binary or ASCII characters, is a（　　）source.

（2）Because most communication systems are digital systems presently, whether a source is an analog or （　　）, its output must be converted to format that can be transmitted（　　）. This conversion of the source output to a digital form is generally performed by source （　　）, whose output may be assumed to be a（　　）of binary digits.

（3）If X is the output of a discrete source, the entropy（　　）represents the（　　）amount of information emitted by the source. A measure of the efficiency of a source-encoding method can be obtained by comparing the average number of binary digits per output（　　）from the source to （　　）.

（4）In any case, it is always more（　　）to encode blocks of symbol instead of encoding each（　　）separately. By making the（　　）size sufficiently large, the average number of binary digits per output letter from the source can be made arbitrarily（　　）the source entropy.

（5）The two famous sources encoding methods are（　　）and（　　）source encoding algorithms, of which the（　　）encoding algorithms has relatively high efficiency.

（6）One type of sample encoding is to represent each discrete amplitude level by a sequence of binary digits. Hence, if we have L levels, we need R=（　　）bits per sample if L（　　）power of 2, or（　　）if R isn't a power of 2.

（7）The major significance of channel capacity formulas is that they serve as（　　）on the transmission rate for（　　）communication over a noisy channel. The fundamental rate that the channel capacity plays is given by the noisy channel coding theorem due to（　　）in 1948.

（8）There exist channel codes that make it possible to achieve（　　）communication, with as small an error probability as desired, if the transmission rate R （　　）, where C is the channel capacity. If （　　）, it's not possible to make the probability of error tend toward（　　）with any code.

（9）The length of a word is the（　　）of elements in the vector and is usually denoted as （　　）. When the alphabet consists of two elements, 0 and 1, the code is a（　　）code and the elements of any code word are called （　　）. When the elements of a code word are selected from an alphabet having q elements ($q > 2$), the code is （　　）.

（10）It is interesting to note that when q is a power of 2, i.e., $q = 2^b$ where b is a positive integer, each q-ary element has an equivalent binary representation consisting of（　　）bits, and, thus, a non-binary code of block length N can be mapped into a binary code of block length $n =$ （　　）.

（11）There are（　　）possible code words in a binary block code of length n. From these code words, we may select $M = 2^k$ code words ($k < n$) to form a code. Thus, a block of k information bits is mapped into a code word of length（　　）selected from the set of $M = 2^k$ code words. We refer to the resulting block code as (n,k) code, and the ratio （　　）is defined to be the rate of the code.

（12）The number of（　　）error patterns is limited by the number of possible syndromes or coset leaders in the standard array. Clearly, to correct t error implies that we have detected （　　）errors. However, it is also possible to detect more than（　　）errors if we compromise in the error correction capability of the code. In general, a code with minimum distance $d_{\min}$ can detect e_d errors and correct e_c errors, where ($e_d + e_c \leqslant d_{\min} - 1$) and（　　）.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（　　）The design of coded modulation for efficient transmission of information may be divided into two basic approaches, the algebraic approach and probabilistic approach.

（2）（　　）The source entropy means the average amount of information emitted by the source, and the process of representing the source output is a process of representing the source output by a sequence of binary digits.

（3）（　　）The output of an analog source is converted to an digital sequence in the first place, and then encoded.

（4）（　　）According to Shannon noisy channel coding theorem, there must exist channel codes and decoders making it possible to achieve reliable communication, with as small an error probability as desired, only if the transmission rate $R \leqslant C$, where C is the channel capacity.

（5）（　　）Both Cyclic block codes and Convolutional Codes belong to probabilistic code.

（6）（　　）If A=($a_{n-1}a_{n-2} \dots a_1a_0$) is a code word of a cyclic code, then $(a_{n-2}a_{n-3} \dots a_0a_{n-1})$, obtained by a cyclic left shift of the elements of A, is also a code word. In fact, all cyclic shifts of A are code words.

（7）（　　）Generally, if a code having q elements, there are 2^q possible code words. A subset of $M = 2^k$ code words may be selected to transmit k bit blocks of information.

（8）（　　）Only one efficient algorithm has been devised for cyclic codes that make it possible to implement long block codes with a large number of code words in practical communication system.

（9）（　　）The set of all weights in a code constitutes the weight distribution of the code. When all the $M=2^k$ code words have equal weight, the code is called a fixed-weight code or a constant-weight code.

（10）（　　）Since the minimum separation between a pair of code words is $d_{\min}$, it is impossible for an error pattern of weight $d_{\min}$ to transform one of these 2^k code words in the code into another code, because this is an undetected error.

（11）（　　）If the actual number of errors is less than $d_{\min}$, the syndrome will have a nonzero weight.

（12）（　　）Error detection may be used in conjunction with an automatic repeat-request (ARQ) scheme for retransmission of the code word.

（13）（　　）The error correction capability of a code is independent of the minimum distance $d_{\min}$.

3.4　课文参考译文　编码技术

3.4.1　信源编码

通信的目的就是将信源的信息传送到目的地。信源的种类繁多，如无线广播系统的信源产生音频信息（语音或声音），而电视广播系统的信源则输出动态的视频图像和音频信息。这类信源输出的信号都是模拟信号，因此称之为模拟信源。对应地，计算机一类输出离散信息的信源（通常为二进制代码或 ASCII 码），故称之为离散信源。

无论信源是离散还是模拟的，现代通信系统以数字系统为主，其传输信号都是数字信号。因此，（模拟）信源输出的信号必须转换为数字形式后才能在数字通信系统中进行传输，这一转换由信源编码完成，其输出通常为二进制代码。

信源编码实际上就是将信源输出表示为二进制代码形式的过程。若离散信源输出为 X，输出信号的平均信息量为熵 $H(X)$，则可通过比较编码后每符号携带的信息量与 $H(X)$ 的差异，来衡量该信源编码方法的效率高低。

可以证明，任何情况下对一个包含若干符号的符号块进行编码比分别对其中每个符号编码的效率要高得多。若符号块足够大，则编码后平均每个符号所对应的二进制代码个数将无限逼近信源的熵 $H(X)$。最为著名的两个信源编码方式是香农–范诺编码法和霍夫曼编码法。下面我们分别通过两个编码实例来说明这两种编码方法。

例3-1　设一个离散独立信源可以输出 8 个独立的消息 A、B、C、D、E、F、G、H，各符号输出的概率空间如下：

X:	A	B	C	D	E	F	G	H
$P(X)$:	0.01	0.27	0.09	0.14	0.05	0.12	0.03	0.29

利用香农–范诺编码法，对该信源编码的具体编码方法及步骤如下：

①　把各消息按出现概率的大小，由大到小排列；

②　将重排的概率序列分成两组，每组概率之和尽可能接近或相等；

③　对得到的两个分组分别按②的方式再次分组，仍使分成的相应组概率和尽可能相等，这时得到 4 个分组；

④ 按③的方式继续分组，直至每个消息都被单独分割出来为止；

⑤ 对每一次划分得到的第一组消息分配 0，第二组消息则分配 1，以此类推；

⑥ 每个消息的二元编码就由它按⑤分得的所有 0、1 序列给定，如表 3-1 所示。

译表 3-1 香农–范诺编码算法实例

符号	概率	第一次	第二次	第三次	第四次	第五次	编码组	码长
H	0.29	0	0				00	2
B	0.27	0	1				01	2
D	0.14	1	0	0			100	3
F	0.12	1	0	1			101	3
C	0.09	1	1	0			110	3
E	0.05	1	1	1	0		1110	4
G	0.03	1	1	1	1	0	11110	5
A	0.01	1	1	1	1	1	11111	5

例 3-2 仍以前面例 3-1 中的离散独立信源为例，其霍夫曼编码过程可分如下 5 步进行。

① 将信源各个消息按其出现概率大小以降序排列；

② 把排列后的两个最小概率对应的消息分成一组，给其中大的（或小的）一个消息分配 0，另一个分配 1，然后把这两个符号合成为一个符号对待，并求出其概率和；

③ 再次将新得到的符号概率与其他尚未处理过的符号概率按由大到小的顺序重新排列，形成一个新的序列；

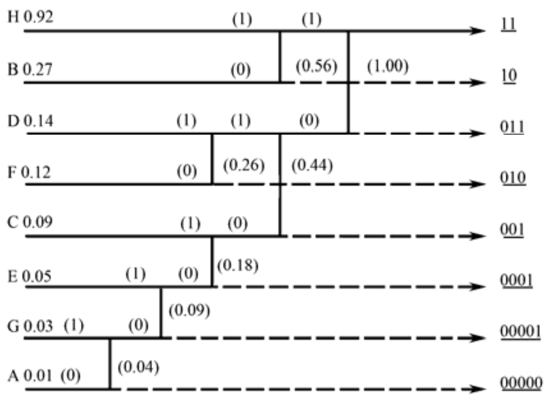
④ 反复重复步骤②，直到所有的概率都已经被联合处理过为止；

⑤ 从下图左边开始，沿着从这个消息为出发点的路线一直到最右边，将遇到的二元数字依次由最低写到最高位所得的二元数字序列，就是所求代码。

例 3-2 的完整霍夫曼编码过程如图 3-1 所示，所得离散信源中各符号的相应霍夫曼编码如表 3-2 所列。

译表 3-2 例 3-2 中离散信源各符号的霍夫曼编码

符号	A	B	C	D	E	F	G	H
码组	00000	10	001	011	0001	010	00001	11



译图 3-1 霍夫曼编码

利用抽样定理，模拟信源的输出被转换为时间离散的抽样序列，再通过量化与编码，成

为数字信号。一种常见的抽样信号编码方式是将每个（量化后的）离散幅度值用二进制数字序列表示出来。设（某量化器的）最大量化值为L，当L是 2 的指数倍时，每个抽样信号都将被表示为$R=\log_2 L$位二进制代码；当L不是 2 的指数倍时，每个抽样信号将被表示为$R=\log_2 L+1$位代码。当信源各符号出现概念确定且不均等时，可采用霍夫曼编码法来提高编码效率。

3.4.2 信道容量和信道编码

1. 简介

香农在 1948 年提出了信道容量表示式，这一公式最重要的意义在于它给出了有噪信道中无误差传输信息的速率上限，并由此奠定了有噪信道编码的理论基础。

当信息传输速率 R 小于信道容量 C，即$R<C$时，必然存在一种信道编解码方式，使得信息可以以无限低的误码率可靠传输。当$R>C$时，则没有一种编码方法能够实现信息的无误码传输。

信息传输中的编码调制方式可分为代数编码法和统计编码法两类。代数编码主要考虑对特定代码的编解码技术，如循环码、卷积码等。统计编码法注重编码信号的统计性能，如代码在信道传输过程中的特性改变及其导致的错误等。

分组码由长度固定的码组组成，一个码组的码元位数就称为该码组的长度，一般用n表示。若每个码元取自包含q个符号的字符表，当$q=2$，即字符表中仅有 0、1 两个字符时，其组成的码组就是二元码，码组中的每个元素就称作二进制位。当$q>2$，其组成的码组就是非二元码。

有意思的是，当q的值是 2 的指数倍时，即$q=2^b$（b为正整数），每个q进制码元都可表示为一个相应的b位二进制码组。因此，长度为N的非二进制分组码必然对应于一个长度$n=b\cdot N$的二进制码组。

长度n的二进制分组码可组成2^n个码组，我们可在其中选择$M=2^k$个码组（$k<n$）构成一个码组。也就是说，k比特信息可映射为一个长度n的码组，其中许用码组有$M=2^k$个，表示为(n,k)分组码，其编码效率 $\eta=k/n$。更一般地，若含有q个码元的码组中有q^n个许用码组，则其子集$M=2^k$个码组可传输k比特信息。

除编码效率η外，还有一个很重要的参数——码重，即码组中每个码组所包含的非零码元个数。一般而言，每个码组中有各种不同码重，所有码重的集合就是该码组的码重分布。特别地，当码组中所有$M=2^k$个码组的码重相等时，这个码组就称作等重码或恒重码。

2. 循环码

循环码属于线性分组码，循环码中任一许用码组经过满环移位后，不论右移或左移，移位位数是多少，所得的新码组仍是许用码组。如：若 A=($a_{n-1}a_{n-2}\ldots a_1a_0$)为一循环码组，则$(a_{n-2}a_{n-3}\ldots a_0a_{n-1})$、$(a_{n-3}a_{n-4}\ldots a_0a_{n-1}a_{n-2})$，等等，都是许用码组。利用循环码的循环特性，可以设计出大量的循环码编、解码电路。

若两个许用码组间的最小距离为$d_{\min}$，则码重为$d_{\min}$的错误图样可转换为2^k个许用码组中的另一个（使接收端将认为是收到另一个许用码组，从而出现错误），从而导致出现无法发现的错误。但若实际发生错误的位数少于最小非零码距$d_{\min}$，则可以发现该一个或多个信道误码。显然，(n,k)分组码能发现（$d_{\min}-1$）个错误，并可与自动重发请求 ARQ 纠错模式

相结合，在发现（接收信码）错误后请求（发送端）重新发送（该）码组。

一个码组的纠错能力与其最小码距 $d_{\min}$ 有关，但其可纠正的错误数还与码组的错误图样以及码组排列有关。一个较简便的判断 (n,k) 码的纠错能力的思路是把 2^k 个许用码组视作 n 维空间中的点，每个码组都是一个以汉明距 t 为半径的圆心，运算符 $\lfloor x \rfloor$ 表示 x 的最大整数取值，则 2^k 个圆彼此均不相交或相切的 t 的最大取值为 $t = \frac{1}{2}\lfloor d_{\min} - 1 \rfloor$，每个圆内则是所有距离禁用码组不大于 t 的许用码组。这意味着当 (n,k) 的最小码距为 $d_{\min}$ 时，它可以纠正 $t = \frac{1}{2}\lfloor d_{\min} - 1 \rfloor$ 个错误。

换句话说，考查某编码的纠错能力时，要纠正 t 个错误，则至少应能发现 t 个或更多个错误。总之，最小码距为 $d_{\min}$ 的码组可发现 e_{d} 个错误，并纠正 e_{c} 个错误。其中

$$e_{\mathrm{d}} + e_{\mathrm{c}} \leqslant d_{\min} - 1 \tag{3-1}$$

且

$$e_{\mathrm{c}} \leqslant e_{\mathrm{d}} \tag{3-2}$$

Unit 4　Multiple Access

4.1　Text

4.1.1　Multi-user Communication System

In modern communication system, the available bandwidth is shared by multiple users in order to improve spectrum efficiency, this kind of system is multi-user communication system.

It is instructive to distinguish among several types of *multi-user communication systems*. One type is a multiple access system in which a large number of users share a common communication channel to transmit information to a receiver. Such a system is depicted in Figure 4-1 (a). The common may be the *up-link in a satellite communication system*, or a cable to which are connected a set of terminals that access a central computer, or some frequency band in the radio spectrum that is used by multiple users to communicate with a radio receiver.

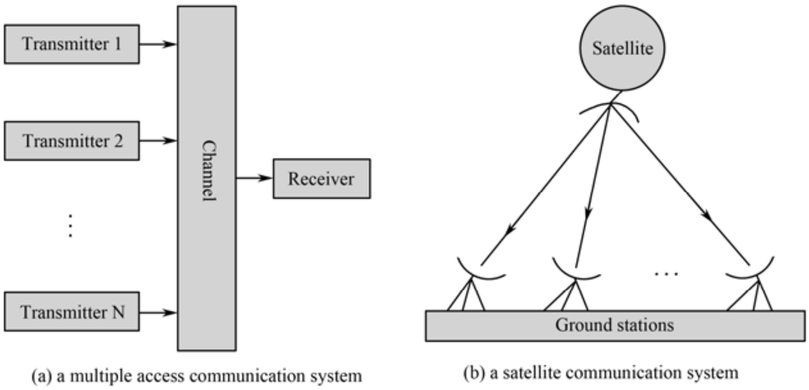


Figure 4-1　Multi-users communication

A second type of multi-user communication system is a broadcast network in which a signal transmitter sends information to multiple receivers as depicted in Figure 4-1(b).

In multi-user communication systems, signals from multiple terminals are multiplexed into a large aggregate data stream. At the receiving end, the data stream is separated the right way around. Techniques to accomplish this task are multiple-access or channel multiplexing.

Of the various channels available for digital communications, telephone channels are by far the most widely used. Such channels are characterized as band-limited linear filters. This is certainly the proper characterization when ***frequency-division multiplexing*** (***FDM***) is used as a means for establishing channels in the telephone network. Recent additions to the telephone network employ ***pulse-code modulation*** (***PCM***) for digitizing and encoding the analog signal and ***time-division multiplexing*** (***TDM***) for establishing multiple channels. Nevertheless, filtering is still used on the analog signal prior to sampling and encoding.

With multiple-access, multiple users from different locations share the common transmission

medium. To accomplish the communication between multiple users, multiple access is designed to be capable of identifying various users, so the signal segmentation theory is employed, i.e. signal from each user is assigned a unique feature, namely “address” and all the user signals are identified by their unique addresses. At the receiving end, signals are separated respectively according to their unique addresses so that the transmission of multiple signals is completed without interference.

Multiple access techniques form the basis for current and future wire-line and *wireless communication networks*, such as *satellite networks*, *cellular and mobile communication networks*, and underwater acoustic network.

According to the way that multiple users share the available bandwidth, there are three basic kinds of multiple access techniques, the frequency-division multiple access (FDMA), the time-division multiple access (FDMA) and the code-division multiple access (CDMA).

4.1.2 Frequency-Division Multiple Access

In general, there are several different ways in which multiple users can send information through the communication channel bandwidth into a number, say *N*, of frequency non-overlapping sub-channels, as shown in Figure 4-2, and to assign a sub-channel to each user upon request by the users. This method is generally called frequency-division multiple access (FDMA), and is commonly used in wire-line channels to accommodate multiple users for voice and data transmission.

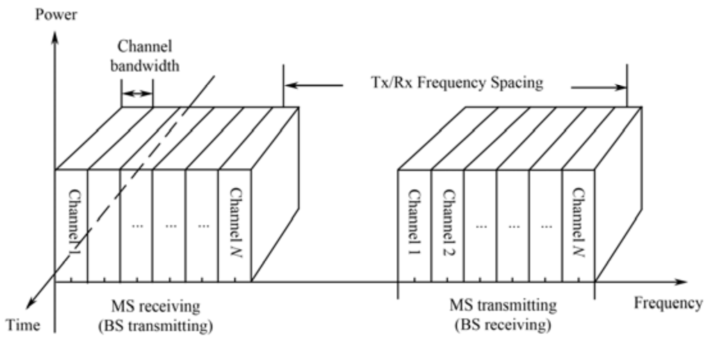


Figure 4-2 Channel division scheme in FDMA

With the development of communication technology, there is an urgent need for more frequency spectrum. FDMA is an efficient approach to solve the problem of bandwidth shortage. In Figure 4-2, there are *N* signals for multiplexing. Firstly, every signal is modulated by a carrier wave with different frequency from each other. Secondly, the modulated signals are multiplexed through an adder and transmitted. FDMA is one of the fundamental techniques employed in analog carrier communication, microwave communication and satellite communication. The 800MHz AMPS in North America and 900MHz TACS in Europe and China are both based on FDMA technique.

In a satellite communication system as shown in Figure 4-3, there are four ground stations on earth. The whole bandwidth is divided into four non-overlapping sub-channels and assigned to corresponding ground stations respectively. At the receiving end, all the signals received are

identified by their unique carrier frequency, e.g. if f'_B is received at station A, we know it is transmitted from Station B.

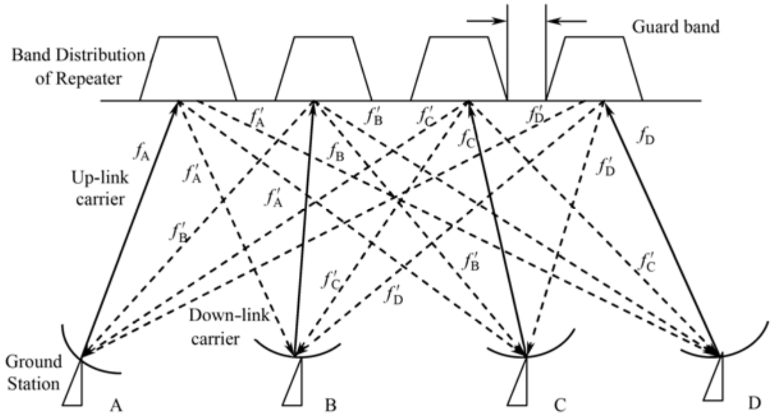


Figure 4-3 FDMA Mechanism

4.1.3 Time-Division Multiple Access

Another method for creating multiple sub-channels for multiple access is to subdivide the duration T_f, called the frame duration, into, say, N non-overlapping subintervals, each of duration T_f/N, is assigned to one user as shown in Figure 4-4.

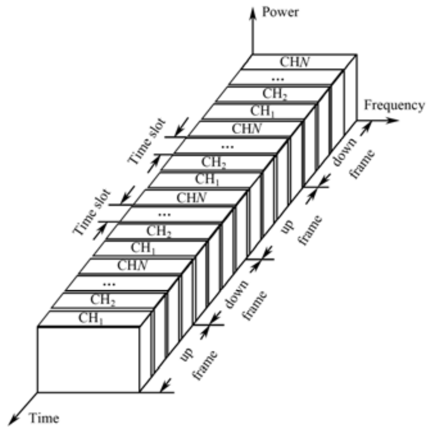


Figure 4-4 Channel division scheme in TDMA

In other words, users in TDMA communication system are distinguished from each other by their different time intervals. Each user who wishes to transmit information is assigned to a particular ***time slot*** within each frame. This multiple access method is called ***time-division multiple access*** (TDMA) and is frequently used in data and digital voice transmission.

In TDMA system, channels are sub-divided into several non-overlapping time slots and in each slot, only one user is allowed to either to transmit or receive. At the receiving end, signal from one particular user can be extracted based on its unique time sequence.

TDMA is mainly used to transmit TDM signals. Typical ones are TDM/PCM/PSK/TDMA systems. TDMA forms the fundamental technique in digital communications such as GSM 900 in China.

We observe that in FDMA and TDMA, the channel is basically portioned into independent single-user sub-channels. In this sense, the communication system design methods that we have described for single-user communication are directly applicable and no new problems are encountered in a multiple access environment, except for the additional task of assigning available

channels to users.

4.1.4 Code-Division Multiple Access

When the information transmission from a single user is separated by periods of no transmission, where these periods of silence may be greater than the periods of transmission, how about the FDMA or TDMA? In fact, this case is generally occurring with users at various terminals in a computer communication network that contains a central computer, and also with users in ***mobile cellular communication systems*** carrying digitized voice, since speech signals typically contain long pauses. In such environment where transmission from various users is low-duty-cycle, FDMA and TDMA become inefficient because a certain percentage of the available frequency slots or time slots assigned to users do not carry information.

As an alternative to FDMA and TDMA, CDMA is introduced to us to allow more than one user to share a channel or sub-channel by use of spread spectrum signals. In this method, each user is assigned a unique code sequence that allows the user to spread the information signal across the assigned frequency band. Thus signals from the various users are separated at the receiver by cross-correlation of the received signal with each of the user pseudo-random sequence. This multiple access method is called ***code-division multiple access*** **(*CDMA*)**.

In CDMA communication system as shown in Figure 4-5, users may access channels in a random manner. The transmitting signal overlaps both in time and frequency, and occupies the whole channel bandwidth both in frequency and time domain as shown in Figure 4-5. The demodulation of multiple signals at the receiving end relies on the pseudo-random code sequence from spread spectrum modulation. As a result, CDMA is also called ***spread-spectrum multiple access*** (***SSMA***).

Spread spectrum technique forms the fundamentals of the CDMA system. The bandwidth of carrier signal is several orders of magnitudes (100 to 1000) greater than the signal bandwidth. CDMA technique is grouped as direct sequence (DS) spread spectrum, frequency hopping (FH) spread spectrum, chirp modulation, time-hopping (TH) and so on. Among these spread spectrum techniques, DS and FH are most common ones and chirp modulation is mainly used in radar system.

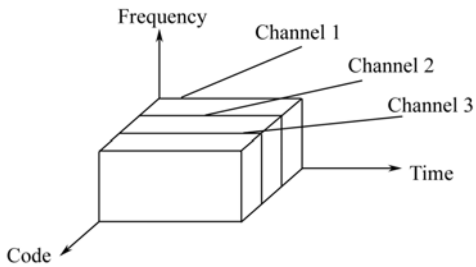


Figure 4-5 Code-division multiple access

4.1.5 Spread Spectrum Modulation

Spread spectrum signals used for the transmission of digital information are distinguished by the characteristic that their bandwidth W is much greater than the information rate R in bits/s.

That is, the ***bandwidth expansion factor*** $B_e = W/R$ for a spread spectrum signal is much greater than unity. The large redundancy inherent in spread spectrum signals is required to overcome the several levels of interference that are encountered in the transmission of digital information over some radio and ***satellite channels***. Since coded waveforms are also characterized by a bandwidth expansion factor greater than unity and since coding is an efficient method for introducing redundancy, it follows that coding is an important element in the design of spread spectrum signals.

A second important element employed in the design of spread spectrum signals is pseudo-randomness, which makes the signals appear similar to random noise and difficult to demodulate by receivers other than the intended ones. This element is intimately related with the application or purpose of such signals. To be specific, spread spectrum signals are used for

① combating or suppressing the detrimental effects of interference due to jamming, interference arising from other users of the channel, and ***self-interference*** due to ***multi-path propagation***;

② hiding a signal by transmitting it at low power and, thus, making it difficult for an unintended listener to detect in the presence of ***background noise***;

③ achieving message privacy in the presence of other listeners.

In combating intentional interference (jamming), it is important to the communicators that the jammer who is trying to disrupt the communication does not have prior knowledge of the signal characteristics except for the overall channel bandwidth and the type of modulation, (PSK, QPSK, etc.) being used.

Interference from the other users arises in the ***multiple-access communication systems*** in which a number of users share a common channel bandwidth. At any given time, a subset of these users may transmit information simultaneously over the common channel to corresponding receivers. Assuming that all the users employ the same code for the encoding and decoding of their respective information sequence, the transmitted signals in this common spectrum may be distinguished from one another by superimposing a different pseudo-random pattern, also called a code, in each transmitted signal. Thus, a particular receiver can recover the transmitted information intended for it by knowing the pseudo-random pattern, i.e., the key, used by the corresponding transmitter. This type of communication technique, which allows multiple users to simultaneously use a common channel for transmission of information, is called ***code division multiple access (CDMA).***

A message may be hidden in the background noise by spreading its bandwidth with coding and transmitting the resultant signal at a low average power. Because of its low power level, the transmitted signal is said to be "covert." It has a low probability of being detected by a casual listener and, hence, is also called a ***low-probability-of intercept (LPI)*** signal.

Message privacy may be obtained by superimposing a pseudo-random pattern on a transmitted message. The message can be demodulated by the intended receivers, who know the pseudo-random pattern or key used at the transmitter, but not by any other receivers who do not have knowledge of the key.

4.1.6 Common Types of Spread Spectrum Modulation

Some spread spectrum systems have employed RF bandwidths 1000 times their information bandwidth. Common spread spectrum systems are of the "***direct sequence***" or "***frequency hopping***" type, or else some combination of these two types (called a "hybrid").

Direct sequence spread spectrum systems are so called because they employ a high speed code sequence, along with the basic information being sent, to modulate their RF carrier.

In a ***Frequency-Hopped*** **(*FH*)** ***spread spectrum*** communication system the available channel bandwidth is subdivided into a large number of contiguous frequency slots. In any signaling interval, the transmitted signal occupies one or more of the available frequency slots. The selection of the frequency slot(s) in each signaling interval is made pseudo-randomly according to the output from a PN generator.

Figure 4-6 illustrates the differences between DS spread spectrum signal and FH spread spectrum signal in time and frequency.

The frequency-hopping rate is usually selected to be either equal to the coded/decoded symbol rate or faster than that. If there are multiple hops per symbol, we have a ***fast-hopped signal***. On the other hand, if the hopping is performed at the symbol rate, we have a ***slow-hopped signal***.

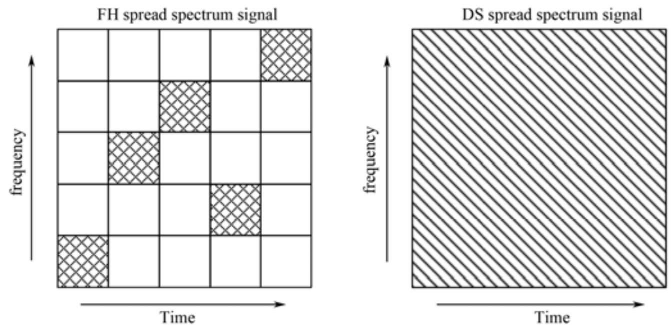


Figure 4-6 differences between DS and FH spread spectrum signal

FH spread spectrum signals are used primarily in digital CDMA communication systems where many users share a common bandwidth. In most cases, a FH signal is preferred over a DS spread spectrum signal because of the stringent synchronization requirements inherent in DS spread spectrum signals. Specially, in a DS system, timing and synchronization must be established to within fraction of the chip interval $T_c \approx 1/W$. On the other hand, in a FH system, the chip interval is the time spent in transmitting a signal in a particular frequency slot of bandwidth $B \leqslant W$. But this interval is approximately $1/B$, which is much larger than $1/W$. Hence the timing requirements in a FH system are not as stringent as in a DS system.

4.1.7 Other Types of Spread Spectrum Signals

There are also"Time Hopped"systems in existence. Time hopped spread spectrum systems have found no commercial application to date. However, the arrival of cheap ***random access memory*** **(*RAM*)** and fast micro-controller chips make time hopping a viable alternative spread

spectrum technique for the future.

However, other methods may be used to introduce pseudo-randomness in a spread spectrum signal. One method, which is analogous to FH, is time hopping (TH). In TH, a time interval, which is selected to be much larger than the reciprocal of the information rate, is subdivided into a large number of time slots. The coded information symbols are transmitted in a pseudo-randomly selected time slot as a block of one or more code words.

For example, suppose that a time interval T is subdivided into 1000 time slots of width $T/1000$ each. With an information bit rate of R bits/s, the number of bits to be transmitted in T is RT. Coding increases this number to RT/R_c bits, where R_c is the coding rate. Consequently, in a time interval of $T/1000$ s, we must transmit RT/R_c bits. If binary PSK is used as the modulation method, the bit rate is $1000R/R_c$ and the bandwidth required is approximately $W = 1000R/R_c$.

Other types of spread spectrum signals can be obtained by combining DS, FH, and TH, such as hybrid DS/FH, which means that a PN sequence is used in combination with frequency hopping. The signal transmitted on a signal hop consists of a DS spread spectrum signal which is demodulated coherently. However, the received signals from different hops combined non-coherently. Since coherent detection is performed within a hop, there is an advantage obtained relative to a pure FH system. However, the price paid for the gain in performance is an increase in complexity, greater cost, and more stringent timing requirement.

Another possible hybrid spread spectrum signal is DS/TH, which isn’t as practical as DS/FH because of an increase in system complexity and more stringent timing requirements.

Technical words and phrases

access ['ækses] *n.* 通路，访问，入门 *vt.* 存取，接近
multi-access 多址接入
wireless ['waiəlis] *adj.* 无线的
wireless communication 无线通信
cellular ['seljulə] *adj.* 蜂窝
mobile ['məubil] *adj.* 移动的
cellular mobile communication networks 蜂窝移动通信系统
acoustic [ə'ku:stik] *adj.* 听觉的；有关 声音的；
underwater acoustic network 水下声讯系统
instructive [in'strʌktiv] *adj.* 有益的，教育性的；指导的；
multi-user communication system 多用户通信系统
up-link 上行链路
sub-channel 子信道
frequency non-overlapping sub-channels 频率非重叠子信道
frequency-division multiple access (FDMA) 频分多址
duration [djuə'reiʃən] *n.* 持续时间，为期
time slot 时隙
TDMA——time-division multiple access 时分多址

segmentation theory　分割理论
AMPS——Advanced Mobile Phone System　先进移动电话系统
TACS——Total Access Communication System　全接入通信系统
single-user communication　单用户通信
silence　['sailəns]　*n.*　静，寂静，沉默，静默；无声的时刻
inefficient　[ini'fiʃənt]　*adj.*　效率低的，效率差的
spread spectrum　频谱扩展，扩频
correlation　[kɑ:ə'leiʃən]　*n.*　相互关系，相关（性）
cross-correlation　互相关性
pseudo　['sju:dəu]　*adj.*　假的，冒充的
pseudo-random sequence　伪随机序列
code-division multiple access (CDMA)　码分多址
demodulation　[di:mɔdju'leiʃən]　*n.*　解调
facilitate　[fə'siliteit]　*vt.*　使容易，使便利，推动，帮助，使容易，促进
spread-spectrum multiple access (SSMA)　扩频多址
direct sequence spread spectrum　直接序列扩频调制
frequency hopping spread spectrum　跳频扩频调制
chirp modulation　线性调频
time-hopping　跳时
expansion　[iks'pænʃən]　*n.*　扩充，发展；开展，膨胀；
inherent　[in'hiərənt]　*adj.*　固有的，内在的，与生俱来的
overcome [əuvə'kʌm]　*vt.*　战胜，克服，胜过，征服；压倒；制服
randomness　*n.*　随意，无安排
intimately　*adv.*　密切地
application　[æpli'keiʃən]　*n.*　应用，运用，施用
specific　[spi'sifik]　*adj.*　详细而精确的，明确的，特殊的
detrimental　[detri'mentl]　*adj.*　有害的
multi-path　多径
background noise　背景噪声
intentional　[ɪn'tenʃənl]　*adj.*　有意图的，故意的
disrupt　[dis'rʌpt]　*v.*　使中断，使分裂，使瓦解，使陷于混乱，破坏
resultant　[ri'zʌltənt]　*adj.*　作为结果而发生的，合成的
covert　['kʌvət]　*adj.*　隐蔽的，偷偷摸摸的
casual　['kæʒjuəl]　*adj.*　偶然的，不经意的，临时的
low-probability-of intercept (LPI)　低拦截率
identical　[ai'dentikəl]　*adj.*　同一的，同样的
generator　['dʒenəreitə]　*n.*　发生器，生成器
initially　[i'niʃəli]　*adv.*　最初，开头
commence　[kə'mens]　*v.*　开始，着手
PN sequence　伪随机信号

frequency-hopped (FH) 跳频(扩频)
accommodate [ə'kɔmədeit] *vt.* 向……提供，容纳
simultaneous [siml'teinjəs] *adj.* 同时的，同时发生的
rectangular [rek'tæŋgjulə] *adj.* 矩形的，成直角的
enhancement [in'hɑ:nsmənt] *n.* 增进，增加
gain [gein] *n.* 增益;
occupy ['ɔkjupai] *vt.* 占，占用，占领，占据
autocorrelation ['ɔ:təukɔri'leiʃən] *n.* 自相关
subdivide ['sʌbdi'vaid] *v.* 再分，细分
contiguous [kən'tigjuəs] *adj.* 邻近的，接近的，毗边的
fast-hopped signal 快跳频（扩频）信号
slow-hopped signal 慢跳频（扩频）信号
primarily ['praimərili] *adv.* 首先，起初，主要地，根本上
preferred [pri'fə:d] *adj.* 首选的
stringent ['strindʒənt] *adj.* 苛刻的，必须严格遵守的（规则）；严厉的；迫切的
time hopping (TH) 跳时（扩频）
hybrid DS/FH 混合直接序列/跳频扩频

4.2 Reading Materials

1. 3G Handsets Offering Free Skype and Home TV Services

Hutchison 3G, in Partnership with Skype, Sling Media, Yahoo!, Nokia, Google, eBay, Microsoft, Orb and Sony Ericsson, are launching a new handset and services package, called the X-Series.

Customers will be able to make unlimited calls from their mobile using Skype, watch their home television via their mobile using Sling, access their home PC remotely using Orb and have access to the best of internet and messaging services from Yahoo!, Windows Live Messenger and Google.

The X-Series from 3 will be priced like fixed line broadband. It will offer use of mobile internet services free at the time of use, for a flat fee，but the roaming may be an issue.

The global launch of the X-Series from 3 is being made possible by the first two handsets that will support this full range of services: the Nokia N73 and the Sony Ericsson W950i.

Canning Fok, Group Managing Director of Hutchison Whampoa, said “This is the Internet as it was meant to be and what people have been waiting for. Mobile broadband is the natural next step for mobile services, extending the full power of the Internet to mobile handsets. By partnering with the leaders of the Internet and the leading handset makers, the X-Series from 3 will give everyone access to more of what they want, when they want it, and however much of it they want, all free when they use it.”

At today’s launch presentation the 3 Group announced global partnerships with major internet brands Sling Media, Orb and Google. These partnerships build on global agreements announced earlier this year with Skype, Microsoft and Yahoo!

An X-Series customer will be able to make and receive unlimited Skype calls with Skype PC users anywhere in the world, and to any other Skype 3 mobile customer. Skype to Skype calls on a 3 mobile will be free.

Your TV where you are

An X-Series customer who purchases a Slingbox will be able to watch anything they are able to watch on their own TV, including their terrestrial TV, Freeview, cable, and satellite TV, at the same time, on their mobile. Slingbox will also let X-Series customers control their home personal video recorder (PVR) to watch shows they have recorded, pause and rewind live TV, or even queue a recording when away from home using their mobile.

Your PC where you are

Using Orb means people can access the digital content that they have stored on their PC at home, including music files, playlists, digital photos and videos, on their X-Series handset. Orb has specifically designed a user interface for X-Series handsets, which will ensure the X-Series customers taking Orb will receive the best user experience.

Messaging

X-Series from 3 will offer customers text instant messages, to or from Windows Live Messenger or Yahoo! Messenger, to another X-Series handset, or a PC. Sending and receiving text instant messages with an X-Series mobile will be free.

The X-Series from 3 will be available in the UK from the 1st December 2006 and in 3's other markets around the world in early 2007.

(Source: cellular news)

2. Pseudo Noise Code in Spread Spectrum System

One way to look at spread spectrum is that it trades a wider signal bandwidth for better signal to noise ratio. Frequency hop and direct sequence are well-known techniques today. The following paragraphs will describe each of these common techniques in a little more detail and show that pseudo noise code techniques provide the common thread through all spread spectrum types.

Frequency hopping is the easiest spread spectrum modulation to use. Any radio with a digitally controlled frequency synthesizer can, theoretically, be converted to a frequency hopping radio. This conversion requires the addition of a pseudo noise (PN) code generator to select the frequencies for transmission or reception.

Most hopping systems use uniform frequency hopping over a band of frequencies. This is not absolutely necessary, if both the transmitter and receiver of the system know in advance what frequencies are to be skipped. Thus a frequency hopper in two meters, could be made that skipped over commonly used repeater frequency pairs. A frequency hopped system can use analog or digital carrier modulation and can be designed using conventional narrow band radio techniques. De-hopping in the receiver is done by a synchronized pseudo noise code generator that drives the receiver's local oscillator frequency synthesizer.

3. ITU opens up huge online resource

Geneva, 10 September 2007 — ITU Standards produced by the Telecommunication Standardization Sector (ITU-T) are now available online without charge. The announcement follows a highly successful trial conducted from January-October 2007, during which some two million ITU-T Recommendations were downloaded throughout the world.

The aim of the trial was to "increase the visibility and easy availability of the output of ITU-T". Offering standards for free is a significant step for the standards community as well as the wider information and communication technologies (ICT) industry. Now, anyone with Internet access will be able to download any of over 3000 ITU-T Recommendations. These are used by equipment manufacturers, telecommunication network operators and service providers throughout the world to drive the information society. The move further demonstrates ITU's commitment to bridging the digital divide by extending the results of its work to the global community.

Mr Malcolm Johnson, Director of ITU's Telecommunication Standardization Bureau (TSB), presented the results of the trial to the 2007 meeting of ITU's Council. He said that not only had the experiment been a success in raising awareness of ITU-T, it would also attract new members. Most importantly, he noted, it had helped efforts to bridge the "standardization gap" between countries with resources to pursue standardization issues and those without. "There has been very positive feedback from developing countries,"said Johnson."Last year exactly 500 ITU-T Recommendations had been sold to developing countries; this year, after allowing free access, they have downloaded some 300 000."

ITU-T Recommendations are developed in a unique contribution-driven and consensus-based environment by representatives of industry and government, with industry providing the most significant technical input.

A strong focus of current standards work is laying the foundations for the next-generation network (NGN). Other key areas include IPTV, ICT in vehicles, cybersecurity, quality of service, multimedia, emergency communications and standards for access, such as VDSL 2 — very high speed digital subscriber line 2, the newest and most advanced standard of DSL broadband wireline communications.

4. Model of Spread Spectrum Digital Communication System

The block diagram shown in Figure 4-7 illustrates the basic elements of a spread spectrum digital communication system with a binary information sequence at its input, at the transmitting channel, and at its output at the receiving end.

The channel encoder and decoder and the modulator and demodulator are basic elements of the system. In addition to these, there are two identical pseudo-random pattern generators, one that interfaces with the modulator at the transmitting end and a second that interfaces with the demodulator at the receiving end. The generators generate a pseudo-random or pseudo-noise (PN) binary-valued sequence, which is impressed on the transmitted signal at the modulator and removed from the received signal at the demodulator.

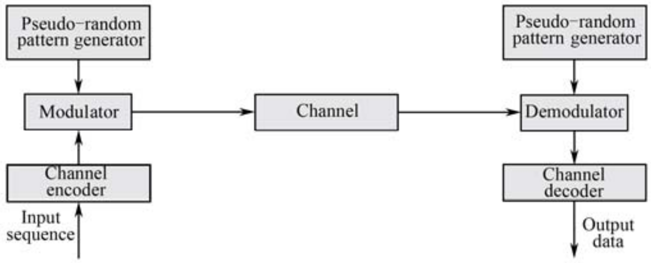


Figure 4-7　a spread spectrum digital communication system model

Synchronization of the PN sequence generated at the receiver with the PN sequence contained in the incoming received signal is required in order to demodulate the received signal. Initially, prior to the transmission of information, synchronization may be achieved by transmitting a fixed pseudo-random bit pattern that the receiver will recognize in the presence of interference with a high probability. After time synchronization of the generators is established, the transmission of information may commence.

When used in conjunction with binary or M-ary (M>2) PSK, the PN sequence generated at the modulator is used in conjunction with the PSK modulation to shift the phase of the PSK signal pseudo-randomly. The resulting modulated signal is called a ***direct sequence*** **(*DS*)** or a ***pseudo-noise*** **(*PN*)** spread spectrum signal. When the pseudo-random sequence selects the frequency of the transmitted signal pseudo-randomly, the resulting signal is called a frequency-hopped (FH) spread spectrum signal.

In the model shown in Figure 4-7, we assume that the information rate at the input to encoder is R bits/s and the available channel bandwidth is W Hz. The modulation is assumed to be binary PSK. In order to utilize the entire available channel bandwidth, the phase of the carrier is shifted pseudo-randomly according to the pattern from the PN generator at a rate W times/s. the reciprocal of W, denoted by T_c, defines the duration of a rectangular pulse, which is called a chip while T_c is called the chip interval. The pulse is the basic element in a DS spread spectrum signal.

The time synchronization of the receiver to the received spread spectrum signal may be separated into two phases. The first step of the synchronization is an initial acquisition phase, and then the second step——tracking phase.

5．RAKE Receiver

Due to complex conditions of mobile station and continuously moving mobile station, multi-path fading of receiving signals will come into being as many reflected signals overlap. Energy of each sub-signal received from one of these multiple paths couldn't meet the requirement of demodulation at receiver. To solve the problem, RAKE receiver is adopted both in base station and in mobile station of CDMA system. Because its action is somewhat analogous to an ordinary garden rake, and, consequently, this receiver coined by Price and Green in 1958 is named "RAKE receiver".

By means of searching and collecting signals from multi-path and then merging them to make signal intensity meet the requirement for demodulation, RAKE receivers can compensate the offset, attenuation and distortion derived from digital signal transmitted over a frequency-selective channel.

Even in worst case, it can ensure the articulation of communication to the utmost extent, enhance the receiving speech to noise ratio and improve performance of system.

Usually, each mobile station receiver uses three RAKE receivers in CDMA system. A set of antenna uses four RAKE receivers in base station.

4.3 Exercises

1．Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）多址接入
（2）无线通信网络
（3）多用户通信系统
（4）频率非重叠子信道
（5）频分多址接入
（6）时隙
（7）移动蜂窝通信系统
（8）码分多址接入
（9）扩频调制信号
（10）无线扩频通信
（11）扩频因子
（12）低拦截率
（13）跳频扩频
（14）卫星信道
（15）多径干扰
（16）背景噪声
（17）多址通信系统
（18）快跳频信号

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）FDMA is a way to subdivide the common channel into more than one（　　）sub-channels for multiple users sending information through one channel, with the entire name of（　　）, and is commonly used in wire-line channels to accommodate（　　）users for voice and data transmission.

（2）Another method for creating multiple（　　）for multiple access is to subdivide the duration T_f into more than one（　　）subintervals assigned to users for information transmission. This multiple access method is called（　　）and briefly described as（　　）.

（3）When the information transmission from a single user is separated by periods of no transmission, where these periods of silence may be greater than the periods of transmission, FDMA and TDMA become（　　）because a certain percentage of the（　　）frequency

slots or（　　）slots assigned to users do not（　　）information.

（4）As an（　　）to FDMA and TDMA, CDMA allows more than one user to share a channel or（　　）by use of（　　）spectrum signals.

（5）In CDMA, each user is assigned a（　　）code sequence allowing user to spread the information signal across the assigned frequency band. Thus signals from the various users are separated at receiver by（　　）of the received signal with each of the user（　　）sequence.

（6）Because a CDMA user accesses channel in a random manner, the signal transmission among the multiple users completely overlaps both in（　　）and （　　）. The demodulation and separation of these signals at receiver is facilitated by the pseudo-random code sequence. So CDMA is also called （　　）.

（7）Spread spectrum signal has a distinct feature that its bandwidth W is much（　　）than the information （　　）R in bits/s. That is, the bandwidth expansion factor B_e=（　　）for a spread spectrum signal is much greater than unity. The large（　　）inherent in spread spectrum signals is required to overcome interference encountered in the transmission of digital information over some radio and satellite channels.

（8）The pseudo-random sequence, which is usually briefly expressed as（　　）sequence, makes the spread spectrum signal similar to random noise and difficult to（　　）by receivers other than the intended ones.

（9）In a multiple-access communication systems, more than one users may transmit information simultaneously over a（　　）channel to corresponding receivers. Assuming that all the users employ the same code for the encoding and decoding of their respective information sequence, the transmitted signals in this common spectrum may be distinguished from one another by superimposing a different（　　）pattern in each transmitted signal. Thus, a particular receiver can recover the transmitted information intended for it by knowing the PN pattern used by the（　　）transmitter. This type of communication technique, which allows multiple users to simultaneously use a（　　）channel for transmission of information, is called code division multiple access (CDMA).

（10）A message may be hidden in background noise by spreading its bandwidth with coding and transmitting the resultant signal at a low average （　　）. Because of its low（　　）level, the transmitted signal is said to be "covert." It has a low（　　）of being（　　）by a casual listener and, hence, is also called a LPI signal.

（11）Message privacy may be obtained by superimposing a pseudo-random pattern on a transmitted message. The message can be demodulated by the（　　）receivers, who know the（　　）pattern used at the transmitter, but not by any other receivers who （　　）have knowledge of the key.

（12）Basic elements of a spread spectrum digital communication system are the channel（　　）; decoder, the modulator and （　　）. In addition, there are two is identical（　　）generators, one at the transmitting end and another at receiving end. The generators generate（　　）sequence, which is impressed on the transmitted signal at the

（　　）and removed from the received signal at the demodulator.

（13）The enhancement in performance obtained from a DS spread spectrum signal through the processing（　　）and coding（　　）can be used to enable many DS spread spectrum signals to occupy the same（　　）bandwidth provided that each signal has its own（　　）PN sequence.

（14）The（　　）among a number of PN code sequences is not easily achieved, especially if the number of PN code sequence required is large. In some applications, the（　　）properties of PN sequences are as important as the autocorrelation properties. Ideally, the PN sequence among users shouldn't be mutually（　　）so that the level of（　　）experienced by any one user from transmissions of other users adds on a power basis.

（15）FH spread spectrum signals are used primarily in digital（　　）communication systems where many users share a common bandwidth. In most cases, a FH signal is preferred over a（　　）spread spectrum signal because of the stringent synchronization requirements inherent in DS signals. In a DS system, timing and synchronization must be established to within fraction of the chip interval $T_c \approx 1/W$. On the other hand, in a FH system, the chip interval is the time spent in transmitting a signal in a particular（　　）of bandwidth $B \leqslant W$. But this interval is approximately $1/B$, which is much larger than（　　）. Hence the timing requirements in a FH system（　　）as stringent as that in a DS system.

（16）Time hopping is a method analogous to FH, in which a（　　）interval is subdivided into a large number of time slots. The coded information symbols are transmitted in a（　　）selected time slot as a block of one or more code words. Suppose that a time interval T is subdivided into 1000 time slots of width $T/1000$ each. With an information bit rate of R bits/s, the number of bits to be transmitted in T is（　　）. Coding increases this number to（　　）bits, where R_c is the coding rate. Consequently, in a time interval of $T/1000$s, we must transmit（　　）bits. If binary PSK is used as the modulation method, the bit rate is（　　）and the bandwidth required is approximately（　　）.

（17）Other types of spread spectrum signals can be obtained by（　　）DS,（　　）and TH, such as hybrid DS/FH, which means that a PN sequence is used in combination with frequency hopping. The signal transmitted on a signal（　　）consists of a（　　）spread spectrum signal demodulated coherently. Since coherent detection is performed within a（　　）, there is an advantage obtained relative to a pure FH system. However, the price paid for the gain in performance is an increase in complexity, greater cost, and more stringent timing requirement.

3. Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（　　）Multiple access techniques form the basis of modern communication systems and communication networks.

（2）（　　）A broadcast network in which a signal transmitter sends information to multiple receivers is a type of multi-point to multi-point communication system.

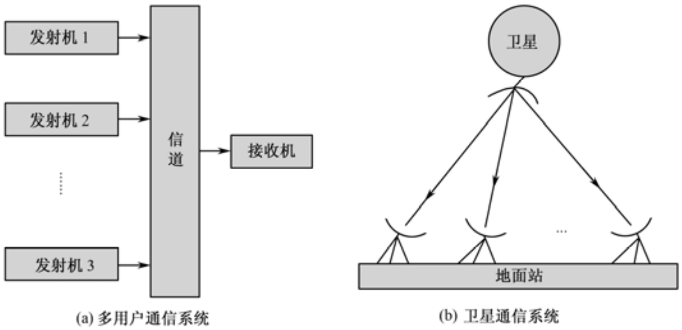
(3)(　　) Frequency-division multiple access can be briefly expressed as FDMA.
(4)(　　) Each subdivided duration T_f/N in TDMA is usually called as frequency slot.
(5)(　　) FDMA and TDMA do not fit the circumstance where transmission from various users is low-duty-cycle because of inefficiency.
(6)(　　) CDMA allows more user to share a common channel than that of FDMA and TDMA.
(7)(　　) In CDMA, each user is assigned a unique code sequence that allows the user to spread the information signal across the assigned frequency band at different time slot.
(8)(　　) One distinct feature of spread spectrum signal is that its bandwidth expansion factor $B_e=W/R$ is a constant.
(9)(　　) The large redundancy inherent in spread spectrum signals is the only reason for its widely used in digital communication system to overcome interference encountered in some radio and satellite channels.
(10)(　　) Interference from other users arises in the multiple-access communication systems in which a number of users share a common channel bandwidth.
(11)(　　) This type of communication technique, which allows multiple users to simultaneously use a common channel for transmission of information, is called CDMA.
(12)(　　) A message may be hidden in the background noise by spreading its bandwidth with coding and transmitting the resultant signal at a high average power.
(13)(　　) Message privacy may be obtained by superimposing a pseudo-random pattern on a transmitted message. The message can be demodulated by the intended receivers, who know the pseudo-random pattern or key used at the transmitter, but not by any other receivers who do not have knowledge of the key.
(14)(　　) The frequency-hopping rate is usually selected to be either equal to the coded/decoded symbol rate or faster than that. If there are multiple hops per symbol, we have a fast-hopped signal. On the other hand, if the hopping is performed at the symbol rate, we have a slow-hopped signal.
(15)(　　) Compared with hybrid DS/FH, DS/TH is used more in practice because of its relatively simple system and less stringent timing requirements.

4.4　课文参考译文　多址接入

4.4.1　多用户通信系统

现代通信系统中，为了提高频率利用率，常常使多个用户信号共享同一个信道进行信息传输，这就是多用户通信系统。

目前有多种类型的多用户通信系统，图 4-1（a）所示的多址系统就是一种多用户通信系统。该系统中的大量用户共用一个信道传送信息到接收机，这个共用信道可以是卫星通信系统的上行链路，或是与接入中心计算机的一组终端相连接的电缆，或是与某一无线接收机相联系供多个用户使用的无线频谱的某个频带。



译图 4-1　多用户通信系统

广播网也是一种常用的多用户通信系统。该网络中，一部发射机发送信息到多个接收机，如图 4-1（b）所示。

多用户通信系统中，在发送端将多路信号合并、接收端再将该合路信号正确分离的技术，就是多址技术。

在各类通信信道中，电话信道最为典型。这类信道具备带限线性滤波器的特征。在电话通信网中通过频分复用（FDM）建立信道。后来，电话通信网使用脉冲编码调制（PCM）对模拟信号进行模数转换和编码，并通过时分复用（TDM）建立信道。不过，在模拟信号采样和编码之前仍进行滤波处理。

多址技术把处于不同地点的多个用户接入一个公共传输媒介，以实现各用户之间的通信。多址技术主要解决用户识别的问题，其理论基础是信号分割理论，即赋予每个信号不同的特征——即“地址”后，按“地址”进行分发。然后在接收端再根据各信号之间的地址差异分离信号，实现互不干扰的通信。

多址接入是现代电缆通信和无线通信网络的核心技术，也是未来通信网的基本技术之一，目前卫星通信、蜂窝移动通信以及水下声讯网络中均使用了多址技术。

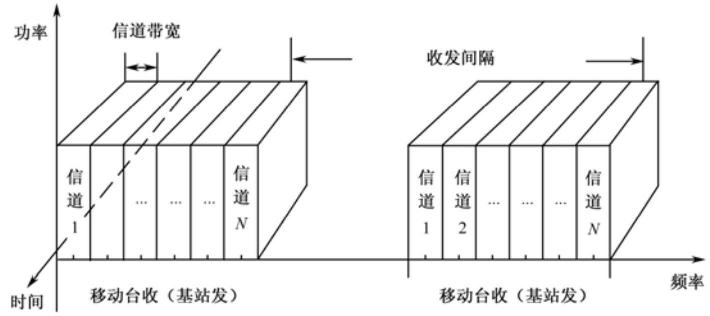
按照实现多个用户接入同一个信道的不同方式，多址技术主要包括三种：频分多址 FDMA、时分多址 TDMA、码分多址 CDMA。

4.4.2　频分多址

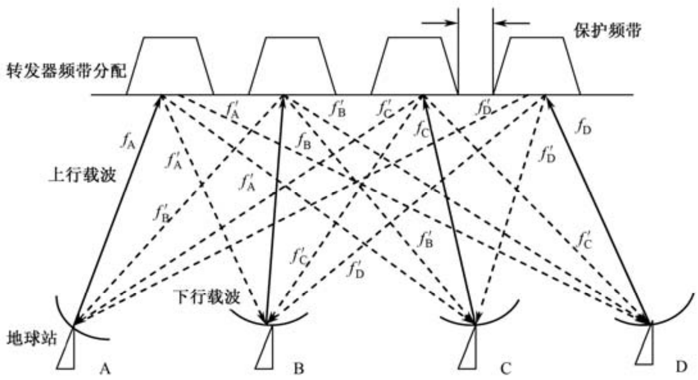
总的来说，使多个用户信号共用一个信道进行传输的多址技术主要有三种。最简单的方法是把可用信道带宽划分为许多个（如 N 个）频率互不重叠的子信道，如图 4-2 所示，并按用户请求把子信道分配给每个用户。这种多址接入方式就是频分多址 FDMA 技术，它主要用于模拟载波通信、微波通信和卫星通信系统的语音和数据传输过程中。

随着通信技术的发展，频率资源日益紧张。频分复用可以提高信道带宽利用率，解决频率或带宽紧缺的问题。图 4-2 中进行复用的信号共有 N 路，每路信号首先分别由不同频率的载波进行调制后，再通过相加器叠加发送。FDMA 是模拟载波通信、微波通信和卫星通信中最基本的技术之一。典型的频分多址方式有北美 800MHz 的 AMPS 体制以及欧洲与我国 900MHz 的 TACS 体制。

图 4-3 所示卫星通信系统中有四个地球站，卫星转发器的整个带宽被分为四个互不重叠的频带后，再分配给相应的地球站作为其发射频带。接收时，各站根据载波的不同频率来识别相应的发射站，如 A 站收到 f_B' 时就知道是 B 站发来的信号。



译图 4-2　FDMA 的信道划分



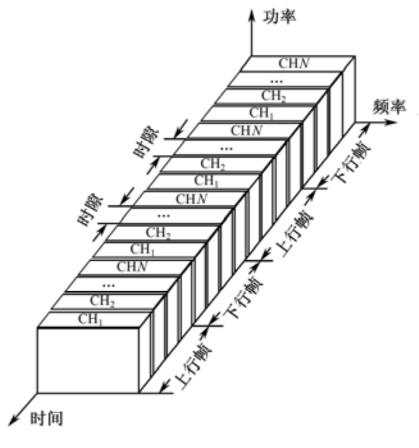
译图 4-3　FDMA 工作方式示意

4.4.3　时分多址

产生多个子信道的另一种方法是把信号的帧持续时间 T_f 划分为 N 个互不重叠的子间隔（时隙），如图 4-4 所示，每间隔持续时间为 T_f/N。每个要发送信息的用户都分配一帧中的一个特定时隙发送信息。这种多址方式叫做时分多址 TDMA 技术，广泛用于数字通信系统的语音和数据传送。

换言之，TDMA 通信系统依据不同的时隙来区分用户。一帧信号被分割成若干个时隙，系统在用户传输信息时为其分配其中一个特定的时隙，这种多址接入方式称为时分多址（TDMA），常用于数据和数字语音传输中。

时分多址技术 TDMA 依靠极其微小的时差，把信道划分为若干不相重叠的时隙，再把每个时隙分配给一个用户专用，在收端就可根据发送各个用户信号的不同时间顺序分别接收不同用户的信号。



译图 4-4　TDMA 的信道划分

时分多址 TDMA 方式主要用来传输 TDM 信号，最典型的是 TDM /PCM/ PSK/TDMA 体制。TDMA 是数字通信中的基本技术，我国的 GSM

900 就采用这一技术。

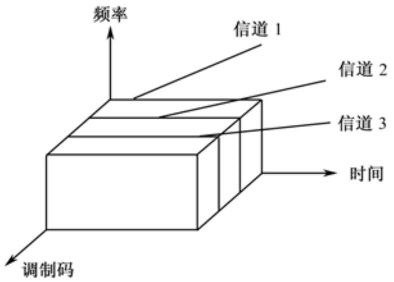
在 FDMA 和 TDMA 中，信息基本上分割为独立的单用户子信道。从这个意义上说，前面介绍的单用户通信系统的设计方法可直接应用于多用户通信，除了增加一项给用户分配可用信道的任务外，在多址环境下不会遇到其他新问题。

4.4.4 码分多址

但实际计算机网络通信中传送的多路信号一般都来自不同的终端，上述单用户信息传送过程中，当用户得到分配子信道、进行信息传送时信号为信息空白（如移动通信系统的语音信号就含有大量的长语音间歇），且该空白数量大于传送的信息量时，FDMA 和 TDMA 由于分配给用户的子信道相对固定，必然导致大量未携带信息的空白子频段或时隙出现，从而使信道利用率降低。

与 FDMA 和 TDMA 不同，码分多址接入（CDMA）方式给每个用户分配一个唯一的扩频序列码，将各路信号用相应的序列码调制后同时在一个信道载频上传输。CDMA 依靠编码的不同来区别各个用户，接收端可利用编码的正交性分离各用户信号，使只有具有相位完全相同的地址码的接收机才能正确解调恢复出原始信号。

CDMA 通信系统中，用户以随机方式接入信道，其信号传输在时间和频率上完全重叠，占用整个信道频段和传输时段，如图 4-5 所示。多路信号在 CDMA 接收端的解调、分路完全依赖于扩频调制的伪随机序列码，所以码分多址 CDMA 技术也常被称作为扩频多址接入 SSMA 技术。



译图 4-5　码分多址的信道划分

CDMA 技术的基础是扩频调制技术，其调制用载波信号的带宽远大于信号带宽（100～1000）倍，主要包括直接序列（DS）扩频、跳频（FH）扩频、线性调频（chirp）、跳时（TH）扩频等几种具体实现技术，其中 DS 和 FH 技术用得较多，而 chirp 技术则主要用于雷达系统。

4.4.5 扩频调制

用于传送数字信息的扩频信号带宽 W 远大于被传送信号的信息速率 R，因此，其扩展因子 $B_e=W/R$ 数值很大，且调制波形也随之具有较大的调制指数。为满足在无线信道或卫星信道上传输信号需克服大量干扰的要求，必须选择适当的信道编码方式，使扩频调制信号携带大量的冗余信息。

此外，设计扩频调制信号的另一个重要因素就是伪随机序列码，它使得被调制信号可具有类似于随机噪声信号的特点，从而不易被接收端非指定接收机解调。这也是调制信号得以普及的主要原因，具体可描述为：

① 降低或消除信道中其他用户信号造成的干扰，以及由于多径传输导致的衰落/畸变等；

② 利用扩频信号的低功率普密度特点增强信号的隐蔽性，使其不易被窃听；

③ 增强信息传输的保密性。

扩频调制能降低恶意干扰的重要原因在于，除被干扰方所使用的信道及其采用的信号调制方式（如 PSK、QPSK 等）外，干扰端几乎完全不可能获知对方的信号特性。

多址接入通信环境中，多个用户共用同一传输频道，任何时刻都有一定数量的用户在同一信道上与其相应目标通信，用户信号之间的彼此干扰也随之增强。假定所有用户对其信息序列采用同一个编、解码形式，则信道中区分各路信号的方式就是对信号各自再按不同的伪随机序列码进行一次调制。这样，只有和发送端具有相同伪随机序列码的接收机方可正确恢复发送端的原始信号。这种允许多个用户同时在同一个信道上传输信号的通信技术就叫做码分多址接入，简称 CDMA。

扩频调制方式利用扩频调制展宽调制信号频带，从而使信号的平均功率极大下降，功率谱密度极低，信号隐藏在背景噪声中，常称之为“被淹没”。正因为此，扩频信号不易被跟踪，是“低拦截率”信号。

此外，由于二次调制所用的伪随机序列，扩频调制信号只能被已知发送端扩频用伪随机序列码的接收机解调，对其他接收端而言只能解调输出随机噪声，故扩频信号保密性高。

4.4.6　常用扩频调制技术

一般地，扩频调制系统采用的射频带宽达信号带宽的 1000 倍，主要为直接序列扩频（DS）和跳频扩频（FH）两种方式，或是这两种扩频方式的组合形式（也称混合型扩频调制）。

直接序列扩频得名于其随基带信号一起用于对 RF 载波调制的高速码序列。

跳频扩频通信系统将传输频段划分为大量的子频段，常称之为频隙。每个信号的传送期间，其载频按 PN 序列发生器产生的 PN 序列在各个频隙中选择一个或多个频隙实现信息传输。

图 4-6 阐述了直接序列扩频信号和跳频扩频信号在信号传输时段和频段上的不同。

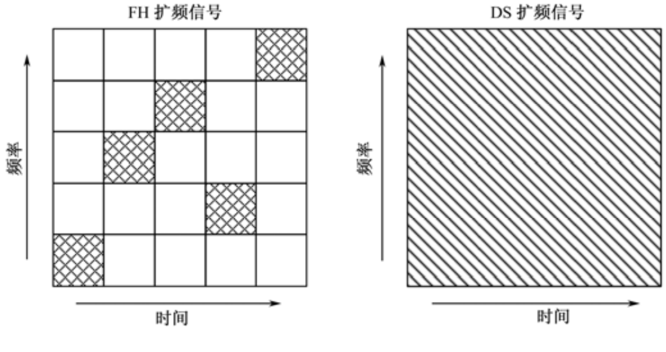


图 4-6　DS 信号与 FH 信号的差异

跳频信号的载波频率跳变速率可以等于或大于信息符号速率。当一个信息符号发送期间载频速率发生了多次跳变时，该跳频扩频信号就是快跳频（F-FH）；若符号发送速率与载频跳变速率相等，则该跳频扩频信号为慢跳频（S-FH）。

跳频信号主要用于多用户共享带宽的 CDMA 移动通信系统。一般而言，跳频信号比直接序列扩频信号在同步性能方面更为优越，因此 FH 信号在 CDMA 系统中的使用多于 DS 信号。采用直接序列扩频方式的 DS 通信系统中，位定时与同步必须在一个码元周期 $T_c \approx 1/W$ 内建立起来。FH 扩频通信系统中，一个码元周期 T_c 为以某一特定频隙（对应带宽为 $B \leqslant W$）传送一个信息符号的时间 $1/B$。由于 $B \leqslant W$，所以 $1/B \geqslant 1/W$。因此，FH 系统对位定时与同步的要求不如 DS 系统高，其实现也就相对容易。

4.4.7　其他扩频信号

尽管直接序列扩频（DS）和跳频扩频（FH）是最常见的两种扩频调制方式，事实上还有一类“跳时”扩频调制系统。虽然目前跳时扩频调制被认为很不经济，随着随机存取存储器 RAM 和高速微控制芯片越来越廉价，跳时（TH）扩频调制有可能投入使用。

与跳频（FH）类似，TH 以远大于信息传输速率倒数的值将传输时间划分为若干个时隙，以快速脉冲的形式在一个或多个时隙上传输信号，而传输使用哪个时隙则由分配给用户的伪随机序列 PN 码决定。

设信号周期 T 被分为 1000 时隙，则每个时隙宽宽为 $T/1000$。若信息速率为 R bits/s，T 时间内发送的比特数就是 RT 。通过编码，该数目增加到 RT/R_c 比特，其中 R_c 为码速率。因此，要在 $T/1000$ 的时间间隔内发送 RT/R_c 个比特。设采用二进制 PSK 调制，则传输比特率就是 $1000R/R_c$，而传输所需带宽约为 W=$1000R/R_c$。

其他类型的扩频信号可以通过同时采用前述 DS、FH 和 TH 三种调制中的两种或多种组合构成，如 DS/FH 混合扩频信号，是直接序列扩频和跳频技术的结合。DS/FH 信号在单个跳频系统中发送的信号是由相干解调的 DS 扩频信号组成的，来自不同跳频系统的接收信号则以非相干方式（如包络或平方律）合并。由于可以在一跳内进行相干检测，所以其性能优于单纯的 FH 系统。然而，这一性能上的提高是以系统复杂性以及成本的增加为代价的，而且它对系统定时和同步精确度的要求更为严格。

另一个可能的混合扩频信号是 DS/TH，但它由于定时精度要求更高、系统更为复杂而不如 DS/FH 实用。

Unit 5 Signal Transmitted in Band-limited Channel

5.1 Text

5.1.1 Signal Transmitted in Band-limited Channel

When the channel is band-limited to bandwidth W, a signal pulse can be designed that allows us to transmit at symbol rates comparable to or exceeding the channel bandwidth W. On the other hand, this transmission must result in ***inter-symbol interference* (*ISI*)** among a number of adjacent symbols.

As a result of the amplitude and delay distortion caused by the non-ideal channel frequency response characteristic $C(f)$, a succession of pulses transmitted through the channel at rates comparable to the bandwidth W are smeared to the point that they are no longer distinguishable as well-defined pulses at the receiving terminal. Instead, they overlap and, thus, we have inter-symbol interference. In addition to linear distortion, signals transmitted through telephone channels are subject to other impairments, specifically nonlinear distortion, frequency offset, phase jitter, impulse noise and thermal noise.

The amount of inter-symbol interference and noise in a digital communication system can be viewed on an oscilloscope. For PAM signals, we can display the received signal $y(t)$ on the vertical input with the horizontal sweep rate set at $1/T$. The resulting oscilloscope display is called an ***eye pattern*** because of its resemblance to the human eye.

For example, Figure 5-1 illustrates the eye patterns for binary and four-level PAM modulation. If there is no ISI, the received signal $y(t)$ is as the same in Figure 5-1(a). On the other hand, if ISI exists, the received signal $y(t)$ should like Figure 5-1(b). For the oscilloscope's horizontal sweep rate is equal to the symbol rate, so the received waveforms of each cyclic period overlap one by one. Thus we get the two eye patterns in Figure 5-1(c) and (d) corresponding with signal waveforms shown in Figure 5-1 (a) and (b) respectively.

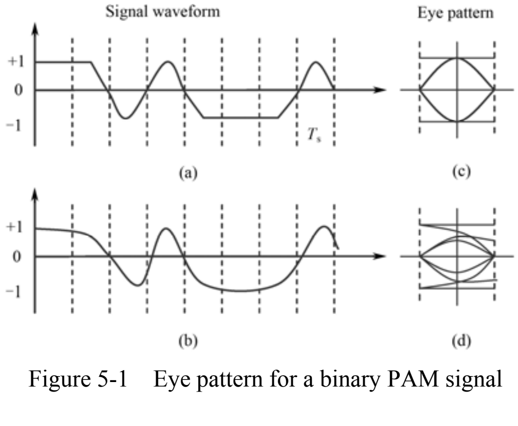


Figure 5-1 Eye pattern for a binary PAM signal

The effect of ISI is to cause the eye to close, thereby reducing the margin for additive noise to cause errors. Figure 5-2 graphically illustrates the effect of inter-symbol interference in reducing the

opening of a binary eye. Note that inter-symbol interference distorts the position of the zero-crossing and causes a reduction in the eye opening. Thus, it causes the system to be more sensitive to a synchronization error.

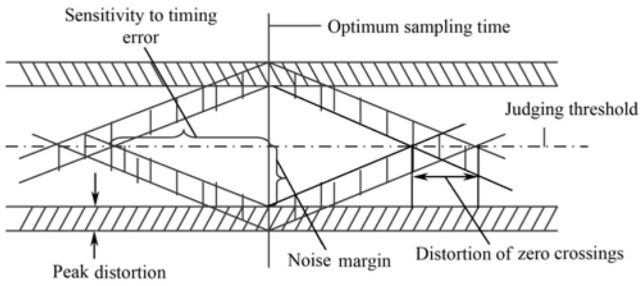


Figure 5-2　Effect of ISI on eye opening

5.1.2　Partial-response Signals and Systems

We assume that the band-limited channel has ideal frequency response characteristics, i.e., $C(f)=1$ for $|f|\leqslant W$. Then the pulse $x(t)$ has a spectral characteristic $X(f)=|G(f)|^2$, where

$$x(t)=\int_{-W}^{W} X(f)\mathrm{e}^{\mathrm{j}2\pi ft}\mathrm{d}f \tag{5-1}$$

The condition for no inter-symbol interference in time domain is as follows:

$$x(t=kT)\equiv x_k=\begin{cases}1 & (k=0)\\ 0 & (k\neq 0)\end{cases} \tag{5-2}$$

The necessary and sufficient condition for the $x(t)$ satisfying (5-2) is that its Fourier transform $X(f)$ satisfies Equation (5-3), which is the Nyquist ISI criterion.

$$\sum_{m=-\infty}^{\infty} X\left(f+\frac{m}{T}\right)=T \tag{5-3}$$

From above, we can know that the smallest value of T for which transmission with zero ISI is possible is T=1/2W. In general, the class of band-limited signals pulses that have the form

$$x(t)=\sum_{n=-\infty}^{\infty} x\left(\frac{n}{2W}\right)\sin\left[2\pi W\left(t-\frac{n}{2W}\right)\right] \tag{5-4}$$

and their corresponding spectra

$$X(f)=\begin{cases}\dfrac{1}{2W}\displaystyle\sum_{n=-\infty}^{\infty} x\left(\frac{n}{2W}\right)\mathrm{e}^{-\mathrm{j}n\pi f/W} & (|f|\leqslant W)\\ 0 & (|f|>W)\end{cases} \tag{5-5}$$

are called ***partial-response signals*** when controlled ISI is purposely introduced by selecting two or more nonzero samples from the set $\{x(n/2W)\}$. The resulting signal pulses allow us to transmit information symbols at the Nyquist rate of $2W$ symbols/s.。

Partial-response signals can reduce or eliminate ISI with relatively high efficiency of channel bandwidth utilization by means of overlapping two or more nonzero samples of $\{x(n/2W)\}$. It is divided into five subclasses for partial-response signals according to the different overlapping manner, i.e., the Ⅰ、Ⅱ、Ⅲ、Ⅳ、Ⅴ partial-response signals. Consequently, the corresponding

systems transmitting various partial-response signals are separately called the Ⅰ、Ⅱ、Ⅲ、Ⅳ、Ⅴ ***partial-response systems***, each of which contains basic elements of a preliminary encoder, a correlative encoder, a sending filter and a receiving filter, etc..

In general, a partial-response signal $x(t)$ can be described more generally as the following (5-6), of which the integer parameters $r_1, r_2, ..., r_N$ are the decisive factor for the partial-response system's type of Ⅰ、Ⅱ、Ⅲ、Ⅳ、Ⅴ by their different value combination. For example, the most widely used two partial-response systems are the Ⅰ and Ⅳ partial-response systems, with which have $(r_0 = r_1 = 1)$ for a Ⅰ partial-response system, and $(r_0 = r_2 = 1) \cap (r_1 = 0)$ for a Ⅳ partial-response systems.

$$x(t) = r_0 \frac{\sin\frac{\pi}{T}\left(t - \frac{T}{2}\right)}{\frac{\pi}{T}\left(t - \frac{T}{2}\right)} + r_1 \frac{\sin\frac{\pi}{T}\left(t + \frac{T}{2}\right)}{\frac{\pi}{T}\left(t + \frac{T}{2}\right)} + \cdots + r_N \frac{\sin\frac{\pi}{T}\left(t + \frac{2N-1}{2}T\right)}{\frac{\pi}{T}\left(t + \frac{2N-1}{2}T\right)} \qquad (5\text{-}6)$$

5.1.3 Synchronization in Band-limited Channel

In a digital communication system, the output of the demodulator must be sampled periodically, once per symbol interval, in order to recover the transmitted information. Since the propagation delay from the transmitter to the receiver is generally unknown at the receiver, symbol timing must be derived from the received signal in order to synchronously sample the output of the demodulator.

The propagation delay in the transmitted signal also results in a carrier offset, which must be estimated at the receiver if the detector is phase- coherent.

Symbol synchronization is required in every digital communication system which transmits information synchronously. ***Carrier recovery*** is required if the signal is detected coherently.

1. Carrier Phase Estimation

There are two basic approaches for dealing with ***carrier synchronization*** at the receiver. One is to multiplex, usually in frequency, a special signal, called a pilot signal, which allows the receiver to extract and, to synchronize its local oscillator to the carrier frequency and phase of the received signal. When an un-modulated carrier component is transmitted along with the information-bearing signal, the receiver employs a ***phase-locked loop*** (***PLL***) to acquire and track the carrier component. The PLL is designed to have a narrow bandwidth so that it is not significantly affected by the presence of frequency components from the information-bearing signal.

The second approach, which appears to be more prevalent in practice, is to derive the carrier phase estimate directly from the modulated signal. This approach has the distinct advantage that the total transmitter power is allocated to the transmission of the information-bearing signal.

Squaring loop is a widely used in practice to establish the carrier phase of ***double-sideband suppressed carrier signals*** such as PAM. It should be noted that the output of the frequency divider has a ***phase ambiguity*** of 180° relative to the phase of the received signal. For this reason, the binary data must be differentially encoded prior to transmission and differentially decoded at the receiver.

Costas loop is another method for generating a properly phased carrier for a double-sideband

suppressed carrier signal. This scheme was developed by Costas (1956) and is called Costas loop. As in the Squaring loop, unfortunately, the Costas loop has the same bug, a phase ambiguity of 180º relative to the phase of the received signal. So the binary data must be differentially processed as in the Squaring loop.

2．Symbol Timing Estimation

In a digital communication system, the output of the demodulator must be sampled periodically at the ***symbol rate****,* at the precise sampling time instants. To perform this periodic sampling, we require a clock signal at the receiver. The process of the extracting such a clock signal at the receiver is usually called ***symbol synchronization*** or ***timing recovery***.

Symbol synchronization is one of the most critical functions performed at the receiver of a synchronous digital communication system. We should note that the receiver must know not only the frequency at which the outputs of the matched filters or correlators are sampled, but also where to take the samples within each symbol interval. The choice of sampling instant within the symbol interval of duration T is called the timing phase.

Symbol synchronization can be accomplished in one of several ways. In some communication systems, the transmitter and receiver clocks are synchronized to a master clock, which provides a very precise timing signal. In this case, the receiver must estimate and compensate for the relative time delay between the transmitted and received signals. Such may be the case for radio communication systems that operate in the ***very low frequency band*** **(*****VLF*****, 0～30kHz)**, where precise clock signals are transmitted from a master radio station.

Another method for achieving symbol synchronization is for the transmitter to simultaneously transmit the clock frequency $1/T$ or a multiple of $1/T$ along with the information signal. The receiver may simply employ a narrowband filter tuned to the transmitted clock frequency and, thus, extract the clock signal for sampling. This approach has the advantage of being simple to implement. There are several disadvantages, however. One is that the transmitter must allocate some of its available power to the transmission of the clock signal. Another is that some small fraction of the available channel bandwidth must be allocated for the transmission of the clock signal.

In spite of these disadvantages, this method is frequently used in telephone transmission systems that employ large bandwidths to transmit the signal of many users. In such a case, the transmission of a clock signal is shared in the demodulation of the signals among the many users. Through this shared use of the clock signal, the penalty in transmitter power and in bandwidth allocation is reduced proportionally by the number of users.

A clock signal can also be extracted from the received signal data. There are a number of different methods that can be used at the receiver to achieve ***self-synchronization***.

Technical words and phrases

band-limited　频带受限的

pulse　[pʌls]　*n.*　脉搏，脉冲

exceed　[ik'si:d]　*vt.*（常与 by 连用）超越，胜过

inter-symbol interference (ISI)　码间干扰
adjacent　[ə'dʒeisənt]　*adj.*　邻近的，接近的
filter　['filtə]　*n.*　滤波器，过滤器，滤光器，
response　[ris'pɔns]　*n.*　回答，响应，反应
distinguishable　[dis'tiŋgwiʃəbl]　*adj.*　可辨识的；易察觉的；显然不同的
terminal　['tə:minl]　*n.*　电路接头；[计] 终端
overlap　['əuvə'læp]　*v.*　（与……）交叠
impairment　[im'pɛəmənt]　*n.*　损害，损伤
offset　['ɔ:fset]　*n.*　偏移量，抵消，弥补，分支，平版印刷，胶印
thermal　['θə:ml]　*adj.*　热的，热量的；温度的
oscilloscope [ə'siləskəup]　*n.*　示波器
vertical　['və:tikəl]　*adj.*　垂直的，直立的，顶点的
horizontal　[hɑ:i'zɔntl]　*adj.*　地平线的，水平的
eye pattern　眼图
resemblance　[ri'zembləns]　*n.*　类同之处；相似
cyclic period　周期
margin　['mɑ:dʒin]　*n.*　边缘；盈余
zero-crossing　过零
synchronization　[siŋkrənai'zeiʃən]　*n.*　同一时刻；同步
spectral　['spektrəl]　*adj.*　光谱的
Nyquist condition for zero ISI　奈奎斯特无码间干扰条件
sufficient condition　充分条件
Fourier transform　傅里叶变换
partial-response signal　部分响应信号
parameter　[pə'ræmitə]　*n.*　参数，参量
periodically　[piəri'ɔdikəli]　*adv.*　周期性地，定时性地
interval　['intəvəl]　*n.*　间隔, 距离；时间间隔
propagation　[prɔpə'geiʃən]　*n.*　（声波，电磁辐射等）传播
synchronously　*adv.*　同时地，同步地
phase-coherent　相位相干的
symbol synchronization　码元同步，位同步
carrier recovery　载波恢复，载波提取
carrier synchronization　载波同步
multiplex　['mʌltəpleks]　*adj.*　多元的；复合的；多重的
pilot signal　导频信号
extract　[eks'trækt]　*vt.*　拔出，榨取，提取
phase-locked loop (PLL)　锁相环
prevalent　['prevələnt]　*adj.*　普遍的, 流行的
allocate　['æləkeit]　*vt.*　（与 to 连用）分配，配给；划拨；拨出
squaring loop　平方环

double-sideband suppressed carrier signals　抑制载波的双边带信号
ambiguity　[æmbi'gju:iti]　*n.*　含糊，不明确
phase ambiguity　相位模糊度
differentially decode　差分编码
Costas loop　科斯塔斯环
symbol rate　码元速率
timing recovery　定时信号恢复，提取定时信号
matched filter　匹配滤波器
correlator　['kɔrileitə]　相关器
compensate　['kɔmpenseit]　*v.*　偿还，补偿，付报酬
very low frequency band (VLF, 0～30kHz) 甚低频
simultaneously　[siməl'teinjəsli]　*adv.*　同时地
tune　[tju:n]　*vt.*　调音，调整，拨收，收听
fraction　['frækʃən]　*n.*　小部分，片断，微量；
proportionally 按比例地，相配合地；适当地
self-synchronization　自同步

5.2　Reading Materials

1．Board chairman of Siemens to step down

The board chairman of German industrial giant Siemens said he was stepping down and hoped that a successor could steer the scandal-plagued company into “calmer waters.”

Heinrich von Pierer, the company’s former CEO, will give up his post at the next supervisory board meeting on April 25. His term would otherwise have run until next year, he said in a company statement.

Von Pierer said he was not stepping down to take responsibility for the corruption investigations that have heaped negative publicity on Siemens, a pillar of the German industrial establishment.

Prosecutors are looking into alleged illegal payments to win business overseas, and current and former executives have been questioned.

But he conceded, “I assume that electing a new chairman of the supervisory board will also make a contribution toward taking our company out of the headlines and bringing it back into calmer waters.”

The Munich-based company’s statement said that board member Gerhard Cromme would be nominated to take over the rest of von Pierer’s term. Cromme is the board chairman of steelmaker ThyssenKrupp and heads the government-appointed commission that produced Germany’s corporate governance code.

Siemens has been rocked by investigations in Germany, Italy and Switzerland over money taken from corporate accounts and allegedly used to pay bribes to help land telecommunications deals.

Six current or former Siemens employees, including the ex-head of its telecommunications equipment unit, Thomas Ganswindt, are suspected of committing breach of trust against Siemens in cases stretching back to 2002 by setting up secret funds outside Germany.

（Source: NewsEdge）

2．Huawei, ZTE build new bases in China

China's two leading telecom equipment providers, Huawei Technologies and ZTE Corp., are planning to build additional manufacturing and R&D bases to speed up domestic and global expansion.

Huawei will reportedly spend $517 million on a base in Dongguan, Guangdong. The base, which will be built in phases, will be ready in early 2008. When fully operational, the cluster of factories will account for several billion dollars worth of Huawei's revenue. The company is also building a similar base in Langfang region, which is scheduled for completion in July 2007 and is expected to generate billions in revenue.

ZTE, meanwhile, is building an R&D and manufacturing park in Shenzhen to beef-up its production lines and expand its presence in the mobile phone industry. The park will cover 440,000sqm and employ 15,000 workers.

（Source: telecom.globalsources）

3．3G

3G is an ITU specification for the third generation (analog cellular was the first generation, digital PCS the second) of mobile communications technology. 3G promises increased bandwidth, up to 384 Kbps when a device is stationary or moving at pedestrian speed, 128 Kbit/s in a car, and 2 Mbit/s in fixed applications. 3G will work over wireless air interfaces such as GSM, TDMA, and CDMA. The new EDGE air interface has been developed specifically to meet the bandwidth needs of 3G.

4．UN Secretary-General Ban Ki-moon visits ITU

Geneva, 9 July 2007 — UN Secretary-General Mr Ban Ki-moon visited ITU headquarters on Friday, 6 July during his visit to Geneva in connection with the high-level ECOSOC meeting and the Global Compact Summit. He brought a clear message of support to ITU, which he termed "one of the most powerful organizations in the UN system".

The UN Secretary-General emphasized the need for connectivity without which "the whole world would be in darkness" and we would continue to live in a medieval age without the means of communication. He said the work of ITU will act as a catalyst in reaching the Millennium Development Goals (MDG) by 2015, and pointed to ITU's work in bridging the digital divide, enhancing cybersecurity and strengthening emergency communications for disaster prevention and relief.

Addressing ITU, Mr Ban Ki-moon called for strong ownership and commitment that would encompass a broader vision. He said ITU is providing the basic groundwork for the international community and should contribute to global agendas such as climate change, which would have

long-term implications for the future of humankind. “ITU is one of the very important stakeholders in the area of climate change”, said Mr Ban Ki-moon.

Dr Hamadoun Touré, Secretary-General of ITU, welcomed Mr Ban Ki-moon to ITU, hailing it as a historic visit — the first to ITU by a UN Secretary-General. Appreciating the importance accorded by Mr Ban to the role of information and communication technologies (ICT) in world development, Dr Touré said that the first priority for ITU is to close the digital divide by 2015, in line with the UN’s Millennium Development Goals. “If ITU does not meet the MDG, then no one else will — because ICT is a tool for everyone,” said Dr Touré. Applauding the UN Secretary-General’s support for ITU’s mission, Dr Touré added, “Together we can help the world to communicate. If we do this, the world will be a better place.”

Earlier, Dr Touré laid out the framework for bridging the digital divide and his vision for achieving cyberpeace. He called for all partners to engage in cybersecurity and for a global approach to emergency communications to enhance disaster preparedness and relief.

5．ITU Radiocommunication Assembly approves new developments for its 3G standards

Geneva, 19 October 2007 — The ITU Radiocommunication Assembly took a decision of global importance to include WiMAX-derived technology in the framework of the IMT-2000 set of standards. This agreement paves the way for the deployment of a range of voice, data, and multimedia services to both stationary and mobile devices. Significantly, it opens the door to mobile Internet, catering to demand in both urban and rural markets.

The ITU Radio communication Assembly (RA-07) formally recognized technology derived from IEEE 802.16 by incorporating it as the sixth terrestrial IMT-2000 radio interface. This is the first addition to IMT-2000 since the original five were adopted years ago as part of the 3G radio standards being used globally and significantly pushes the technological envelope of IMT-2000 capabilities.

IMT-2000 — “International Mobile Telecommunications” — is a global standard defined by ITU in a set of interdependent ITU Recommendations, which include the specifications for the radio interfaces of advanced wireless communications systems such as 3G mobile.

An initial application for the IMT-2000 Advanced standard was made at the ITU-R WP8F meeting in Kyoto, Japan, in January this year. The adoption of the latest radio interface was the culmination of tireless effort among administrations, industry and ITU experts.

5.3 Exercises

1．Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）位同步（符号同步）
（2）锁相环
（3）载波同步
（4）科斯塔斯环
（5）双边带抑制载波信号

（6）相位模糊
（7）平方环
（8）自同步
（9）甚低频
（10）载波恢复
（11）码元速率
（12）定时提取
（13）码间干扰
（14）眼图
（15）部分响应信号
（16）第Ⅰ类部分响应系统
（17）匹配滤波器

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）（　　）Synchronization is required in every digital communication system which transmits information synchronously, but（　　）synchronization is required only if the signal is detected coherently.

（2）There are two basic approaches for dealing with carrier synchronization at the receiver. One is to multiplex a special signal, called（　　）signal, that allows the receiver to extract and, to synchronize its（　　）oscillator to the carrier frequency and（　　）of the received signal. Usually, the receiver employs a（　　）to acquire and track the carrier component, which is designed to have a（　　）bandwidth so that it is not significantly affected by the（　　）of frequency components from the information-bearing signal.

（3）In digital communication system, the output of the demodulator must be samples periodically at the（　　）rate at precise（　　）time instants, requiring a（　　）signal at the receiver. The process of the extracting such a clock at receiver is usually called（　　）synchronization or（　　）recovery.

（4）Symbol synchronization is one of the most critical functions performed at receiver of a synchronous digital communication system. The receiver must know not only the（　　）at which the outputs of the matched filters or correlators are sampled, but also where to take the samples within each（　　）. The choice of sampling instant is called the（　　）phase.

（5）In some communication systems, the transmitter and receiver clocks are synchronized to a（　　）clock, which provides a very precise timing signal. In this case, the receiver must estimate and（　　）for the（　　）time（　　）between the transmitted and received signals.

（6）The clock signal can also be（　　）from received signal data. There are a number of different methods that can be used at（　　）to achieve（　　）.

（7）When signal transmitted in band-limited channel with symbol rate exceeding the channel bandwidth W, the transmission must result in（　　）among（　　）symbols.

(8) Telephone channel are characterized as ( ) linear filter with the proper characterization when frequency-division multiplexing is used for establishing channels in the telephone network. In a word, ( ) is always used on the analog signal prior to sampling and ( ).

(9) The amount of inter-symbol interference and noise in a digital communication system can de viewed on an oscilloscope, the oscilloscope display is called ( ) because of its resemblance to human eye.

3. Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

(1) ( ) The propagation delay in the transmitted signal only results in carrier offset.

(2) ( ) Since the propagation delay from the transmitter to the receiver is generally unknown, symbol timing must be derived from the received signal in order to synchronously sample the output of the demodulator.

(3) ( ) Because that symbol synchronization is required in every digital communication system and carrier synchronization is just required when signal is detected coherently, symbol synchronization is much more important than carrier synchronization.

(4) ( ) Costas loop method is another carrier signal recovery approach overcoming the shortage of Squaring loop of phase ambiguity.

(5) ( ) There are two basic approaches for dealing with carrier synchronization at the receiver, the self-carrier synchronization and the external carrier synchronization.

(6) ( ) Radio communication systems, operating in very low frequency band, implements symbol synchronization by means of synchronizing with a master clock, where precise clock signals are transmitted from a master radio station.

(7) ( ) Another method for achieving symbol synchronization is for the transmitter to simultaneously transmit the clock frequency $1/T$ or a multiple of $1/T$ along with the information signal. This approach has several disadvantages, so it is barely used in practice.

(8) ( ) In spite of these disadvantages of self-symbol-synchronization, it is frequently used in telephone transmission systems that employ large bandwidths to transmit the signal of many users.

(9) ( ) There are a number of different methods that can be used at the receiver to achieve self-synchronization.

(10) ( ) The phenomenon of ISI only happens when a signal pulse is transmitted at symbol rates comparable to or exceeding the band-limited channel's bandwidth.

(11) ( ) The amount of inter-symbol interference and noise for PAM signals can de viewed on an oscilloscope's horizontal display with the vertical sweep rate set at the symbol transmitting rate.

(12) ( ) ISI causes the eye of aye pattern to close, distorts the position of the zero-crossing.

(13) ( ) The partial-response systems can be subdivided into Ⅰ、Ⅱ、Ⅲ、Ⅳ、Ⅴ partial-response systems, each of which contains at least a preliminary encoder and a correlative encoder etc.

（14）（　　）The Ⅰ partial-response system is one of the most widely used partial-response systems, with which have $(r_0 = r_2 = 1) \bigcap (r_1 = 0)$ for Niguist formula < 2 >.

5.4　课文参考译文　带限信道中的信号传输

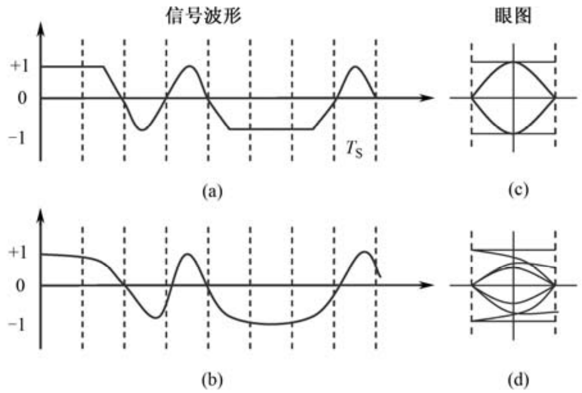
5.4.1　带限信道中的信号传输

对上限带宽为 W 的带限信道，当信号脉冲以不低于 W 的符号速率在信道中传输时，邻近符号间必产生相互干扰，称之为码间干扰 ISI。

信道频率响应特性 $C(f)$的不理想将导致以不低于带宽 W 的符号速率进行传输的信号波形产生延迟和失真，使其叠加到接收端其他信号上，产生码间干扰，影响其抽样判决。除线性失真外，电话信道中传输的信号还易于受到其他损伤，主要是非线性失真、频率偏移、相位畸变、脉冲干扰和热噪声等。

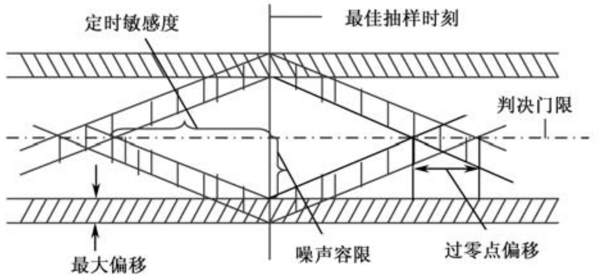
可以用示波器观察到数字通信系统中的码间干扰和噪声影响。如二进制脉冲编码调制 PAM 信号，将接收滤波器的输出信号 $y(t)$ 与示波器垂直输入端连接，并使示波器水平扫描速率与接收码元周期同步，即 $1/T$。这样，示波器上显示的图形就是所谓的眼图，这是因为显示波形很像人的眼睛。

现在我们用图 5-1 来解释这种观察方法。这是一个二元 4 阶 PAM 调制信号的眼图，如果信道中无码间干扰存在，则将得到如图 5-1（a）所示的输出信号波形 $y(t)$；如果有码间干扰，则输出波形 $y(t)$ 将如图 5-1（b）。由于观察用示波器的水平扫描速率与码元速率相等，示波器上显示的将是每个周期的码元波形重叠在一起的波形，即图 5-1 中（c）和（d）图，它们就是分别对应于（a）图、（b）图所示波形的眼图。



译图 5-1　二进制 PAM 信号眼图

ISI 导致眼图中的眼睛闭合，由此降低系统的噪声容限，使误码率上升。图 5-2 阐述了码间干扰 ISI 对二进制信号眼图的影响。可以看出，ISI 干扰使判决门限电平上的眼图过零点向眼睛内部偏移了一段距离，使眼睛的张开程度下降，从而导致系统对判决定时变得敏感。



译图 5-2　码间干扰对眼图的影响

5.4.2　部分响应信号和部分响应系统

设带限信道在带内具有理想频率响应，即当 $|f| \leqslant W$ 时，$C(f)=1$。则脉冲信号 $x(t)$ 的频谱响应必有 $X(f)=|G(f)|^2$，即

$$x(t)=\int_{-W}^{W} X(f)\mathrm{e}^{\mathrm{j}2\pi ft}\mathrm{d}f \tag{5-1}$$

时域脉冲信号 $x(t)$ 无码间干扰条件是：

$$x(t=kT) \equiv x_k = \begin{cases} 1 & (k=0) \\ 0 & (k \neq 0) \end{cases} \tag{5-2}$$

满足式（5-2）的充分必要条件是 $x(t)$ 的傅里叶变换 $X(f)$ 满足式（5-3）所示条件，即奈奎斯特第一准则。

$$\sum_{m=-\infty}^{\infty} X(f+m/T)=T \tag{5-3}$$

由上述定理可知，无码间干扰传输的最小 T 取值为 T=1/2W。综上所述，式（5-4）所示脉冲信号 $x(t)$ 及其频谱 $X(f)$，见式（5-5），就是在带限信道（最大带宽为 W）传输的部分响应信号，其每个码元信号的抽样时刻均被有意叠加了两个或更多个非零的其余 $\{x(n/2W)\}$ 码元波形。这类部分响应信号的最大信息传输速率（常称为奈奎斯特速率）为 2W 符号/秒。

$$x(t)=\sum_{n=-\infty}^{\infty} x(n/2W)\sin[2\pi W(t-n/2W)] \tag{5-4}$$

$$X(f)=\begin{cases} \dfrac{1}{2W}\sum\limits_{n=-\infty}^{\infty} x(n/2W)\mathrm{e}^{-\mathrm{j}n\pi f/W} & (|f| \leqslant W) \\ 0 & (|f|>W) \end{cases} \tag{5-5}$$

通过有目的地叠加两个或多个非零 $\{x(n/2W)\}$ 码元，部分响应信号可降低或消除码间串扰，且保证频带利用率。根据叠加方式的不同，可将部分响应信号分为Ⅰ、Ⅱ、Ⅲ、Ⅳ、Ⅴ类，其相应的传输系统分别为Ⅰ、Ⅱ、Ⅲ、Ⅳ、Ⅴ类部分响应系统，包括预编码器、相关编码器、发送滤波器以及接收滤波器几个部分。

部分响应信号 $x(t)$ 可以更一般地表示为：

$$x(t)=r_0\frac{\sin\frac{\pi}{T}\left(t-\frac{T}{2}\right)}{\frac{\pi}{T}\left(t-\frac{T}{2}\right)}+r_1\frac{\sin\frac{\pi}{T}\left(t+\frac{T}{2}\right)}{\frac{\pi}{T}\left(t+\frac{T}{2}\right)}+\cdots+r_N\frac{\sin\frac{\pi}{T}\left(t+\frac{2N-1}{2}T\right)}{\frac{\pi}{T}\left(t+\frac{2N-1}{2}T\right)} \tag{5-6}$$

其中$r_1,r_2,\cdots,r_N$为加权系数，分别取整数，根据它们的不同取值组合，可确定相应的部分响应系统类型。最常用的是第Ⅰ类和第Ⅳ类，当 $r_0=r_1=1$ 时就是第Ⅰ类部分响应系统；当 $r_0=1$、$r_1=0$、$r_2=1$ 时，则是第Ⅳ类部分响应系统。

5.4.3 带限传输时的同步

数字通信系统中，接收机解调输出的信号必须通过定时抽样、再生，才能正确恢复原始信息。由于信号从发送到接收的过程中产生的时延是不确定的，为与解调再生同步，接收端产生的定时信号必须与收到的信号同频同相。

此外，在要求相位同步的系统中，还需要考虑由于传输时延可能导致的载波衰减。

符号同步是所有同步传输数字通信系统必不可少的，而载波同步则只用于数字通信系统在信号连续传输的情况。

1．载波同步及其相位跟踪

接收端载波同步方式主要有两种。一种是插入导频法，就是在发送端插入一个或几个携带载频信息的导频信号，使已调信号的频谱加入一个小功率的载频频谱分量，接收端只需将它与调制信号分离开来，便可从中获得载波信号。这个额外插入的频谱分量就是导频信号。插入导频法一般分频域插入和时域插入两种，但频域插入法使用更为普遍。

第二种载波提取的方法是直接提取法，它比插入导频法更为常见。直接法不在发送端另外发送同步信号，而是由接收端设法直接从收到的调制信号中提取载波信号。这一方法最主要的优点在于其发送功率完全用于调制信号传送，所以效率较高。

平方环法常用于抑制载波双边带调制信号（如 PAM 信号）的载波提取。但这一方法必须注意其提取的载波信号与接收的载波信号存在 180°的相位模糊，(即两者之间要么同相，要么反相)。因此，发送端输入信号以及接收端接收信号必须分别在发送调制及接收解调前进行差分编码。

科斯塔斯（Costas）环法由科斯塔斯 1956 年提出并由此得名，常用于抑制载波双边带调制信号的载波恢复。科斯塔斯环法与平方环法一样，也存在相位模糊的问题。因此，其信号也必须在发送端调制及接收端解调前分别进行差分编码。

2．码元同步

数字通信中，接收端必须与发送端步调一致。若发送端在发送信息码元的同时发送位定时脉冲序列，使接收端能利用此定时脉冲进行正确的取样判决。这个接收端提取定时脉冲序列的过程就是位同步，也叫做码元同步。

位同步对数字通信系统而言是至关重要的，其接收端不单要获得匹配滤波器或相关器输出信号的频率，还必须确知每个码元的起止时刻。码元持续期间所选择的取样时刻 T 常被称作位定时时刻。

有几种实现位同步的方法。有些通信系统甚低频（VLF, 0～30kHz）无线通信系统，收

发两端的时钟均同步于提供高精度时钟信号的主时钟，接收端不断测算并补偿接收信号由于时延造成的时钟误差。

另一种实现位同步的方法是外同步法，即发送端在发送信号的同时发送频率为 $1/T$ 或 $1/T$ 整数倍的时钟信号，接收端只需通过一个相应的窄带滤波器即可提取时钟信号。该方法最大的优点是易于实现，但有几点不足：一是单独发送的时钟信号占用了部分发送功率；此外，二是必须为此时钟发送留出一定的传输带宽，降低了系统效率。

尽管存在这些缺点，外同步法在多用户宽带电话传输系统中使用较多，这是因为这种情况下，接收端多个用户分摊共用时钟信号，从而降低了时钟信号传输所占用的功率和带宽。

时钟信号也可从接收端收到的信号中提取，这就是自同步法，它又多种不同的具体实现方法。

Unit 6 2G cellular Communication System

6.1 Text

6.1.1 Cellular in Communication Systems

The cellular concept was a major breakthrough in solving the problem of spectral congestion and user capacity. It offered very high capacity in a limited spectrum allocation without any major technological changes. The cellular concept is a system-level idea which calls for replacing a single, high power transmitter (large cell) with many low power transmitters (small cells), each providing coverage to only a small portion of the service area. Each base station is allocated a portion of the total number of channels so that all the available channels are assigned to a relatively small number of neighboring base stations. Neighboring base stations are assigned different groups of channels so that the interference between base stations (and the mobile users under their control) is minimized.

As the demand for service increases (i.e., as more channels are needed within a particular market), the number of base stations may be increased (along with a corresponding decrease in transmitter power to avoid added interference), thereby providing additional radio capacity with no additional increase in radio spectrum. This fundamental principle is the foundation for all modern wireless communication systems, since it enables a fixed number of channels to serve an arbitrarily large number of subscribers by reusing the channels throughout the coverage region. Furthermore, the cellular concept allows every piece of subscriber equipment within a country or continent to be manufactured with the same set of channels so that any mobile may be used anywhere within the region.

Cellular radio systems rely on an intelligent allocation and reuse of channels throughout a coverage region. Each cellular base station is allocated a group of radio channels to be used within a small geographic area called a cell. Base stations in adjacent cells are assigned channel group which contains completely different channels than neighboring cells. The base station antennas are designed to achieve the desired coverage within the particular cell. By limiting the coverage area to within the boundaries of a cell, the same group of channels may be used to cover different cells that are separated from one another by distances large enough to keep interference levels within tolerable limits. The design process of selecting and allocating channel groups for all of the cellular base stations within a system is called frequency reuse or frequency planning.

A cellular telephone system provides a wireless connection to the PSTN for any user location within the radio range of the system. Cellular radio systems provide a very high quality service that open comparable to that of the landline telephone systems. High capacity is achieved by limiting the coverage of each base station transmitter to a small geographic area called a cell so that the same radio channels may be reused by another base station located some distance away. A sophisticated switching technique called a handoff enables a call to proceed uninterrupted when the user moves from one cell to another.

The mobile station contains a transceiver, an antenna, and control circuitry, and may be mounted in a vehicle or used as a portable hand-held unit. The base stations consist of several transmitters and receivers which simultaneously handle full duplex communications and generally have towers which support several transmitting and receiving antennas. The base station serves as a bridge between all mobile users in the cell and connects the simultaneous mobile calls via telephone lines or microwave links to the MSC. A typical MSC handles 100000 cellular subscribers and 5000 simultaneous conversations at a time, and accommodates all billing and system maintenance functions, as well.

Communication between the base station and the mobiles is defined by a standard common air interface (CAI) that specifies four different channels. The channels used for voice transmission from the base station to mobiles are called the forward voice channels (FVC), and the channels used for voice communication from mobiles to the base station are called reverse voice channels (RVC). The two channels responsible for initiating mobiles calls are the forward control channels (FCC) and the reverse control channels (RCC).

All cellular systems provide a service called roaming. This allows subscribers to operate in service areas other than the one from which service is subscribed. When a mobile enters a city or geographic area that is different from its home service area, it is registered as a roamer in the new service area. This is accomplished over the FCC. Every several minutes, the MSC issues a global command over each FCC in the system, asking for all mobiles which are previously unregistered to report their MIN and ESN over the RCC. New unregistered mobiles in the system periodically report back their subscriber information upon receiving the registration request, and the MSC then uses the MIN/ESN data to request billing status from the home location register (HLR) for each roaming mobile. Once registered, roaming mobiles are allowed to receive and place calls from that area, and billing is routed automatically to the subscriber’s home service provider.

6.1.2 Brief Introduction of GSM

***Global System for Mobile Communications* (*GSM*)** mobile communication system is one of the most popular communication systems throughout the world. The term GSM usually means the GSM standard and protocols in the frequency spectrum around 900MHz. There is also DCS1800 - GSM protocols but at different air frequencies around 1800 MHz - and in the United States, where spectrum for ***Personal Communication Services* (*PCS*)** was auctioned at around 1900MHz being named GSM1900. As a result of this, the original and most widely-used GSM frequency implementation is also becoming known as GSM900, and DCS1800 is also known as GSM1800. However, although the physical frequencies used differ, the protocols and architecture remain the same. The basic parameters of GSM are listed in Table 6-1.

Table 6-1 Basic parameters of GSM system

	GSM900	GSM1800	GSM1900
Emission Frequency (MHz)	890～915	1710～1785	1850～1910
Receiving Frequency (MHz)	935～960	1805～1880	1930～1990
Multiplexing	FDD		

（续）

	GSM900	GSM1800	GSM1900
Channel spacing (kHz)	200		
Portable TX power, maximum / average (mW)	1000 / 125		
Power control, handset and BSS	Yes		
Speech coding and rate (Kbps)	RPE-LTP / 13		
Channel rate (Kbps)	270.833		
Channel coding	Rate 1/2 convolutional code		
Frame duration (ms)	4.615	374	299
Modulation Mode	GMSK		
Radius of Cell (km)	<35	<4	<4
Motion Velocity (km/hour)	250	125	125

The emitting power of base station is 500 w per carrier wave. There are five values of the mobile station's emitting power to be chosen, namely 0.8w、2w、5w、8w and 20w. The coverage radius of a small district in GSM 900 is 35km in maximum and 500m in minimum.

6.1.3 Architecture of GSM System

Figure 6-1 shows a simplified diagram of the GSM TDMA format and the structure of the normal burst, which has a throughput after coding of 22.8 Kbps, and offers ***full-rate*** voice at a net bit rate of 13 Kbps and data at up to 8.6 Kbps. GSM has also specified a ***half-rate*** service by time-multiplexing two users onto the TDMA structure. This service offers a gross bit rate of 11.4 Kbps, and data at 4.8 Kbps.

There are three basic elements in a GSM mobile communication system, the ***switching subsystem***, the ***base station subsystem*** and other networks, which is shown in Figure 6-2.

The major functional entities of a GSM system in Figure 6-2 are briefly explained as follows.

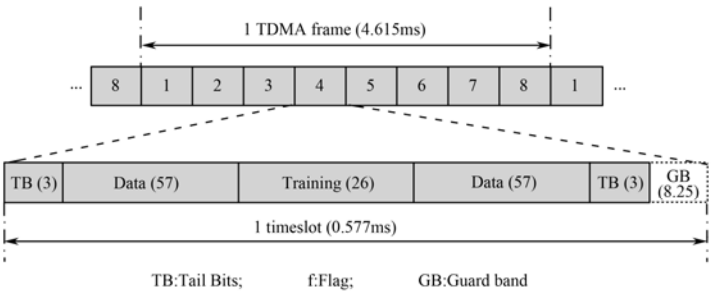


Figure 6-1 GSM TDMA Structure

- MS—***Mobile Station.*** The MS is the physical equipment used by a subscriber, most often a normal ***hand-held cellular telephone.***
- BTS—***Base Transceiver Station***. The BTS comprises the radio transmission and reception devices, and also manages the signal processing related to the air interface.
- BSC—***Base Station Controller.*** The BSC manages the radio interface, mainly through the allocation, release and handover of radio channels.

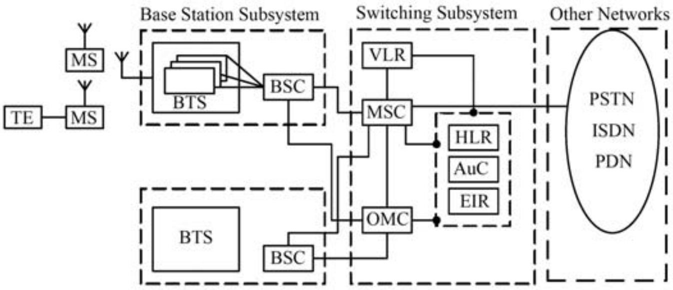


Figure 6-2 Architecture of GSM System

- BSS—***Base Station Subsystem***. The BSS consists of a BSC and one or more BTSs.
- MSC—***Mobile Switching Centre.*** The MSC is basically an ISDN-switch, coordinating and setting up calls to and from MSs.
- VLR—***Visitor Location Register***. The VLR contains all the subscriber data, both permanent and temporary, which are necessary to control a MS in the MSC's coverage area. The VLR is commonly recognized as an integral part of the MSC, rather than a separate entity.
- AuC—***Authentication Centre.*** The AuC database contains the subscriber authentication keys and the algorithm required to calculate the authentication parameters to be transferred to the HLR.
- HLR—***Home Location Register***. The HLR database is used to store permanent and semi-permanent subscriber data; as such, the HLR will always know in which location area the MS is (assuming the MS is in a coverage area), and this data is used to locate an MS in the event of a MS terminating call set-up.
- EIR—***Equipment Identity Register.*** The EIR database contains information on the MS and its capabilities. The ***IMEI*** **(*International Mobile Subscriber Identity*)** is used to interrogate the EIR.
- GMSC—***Gateway Mobile Switching Centre.*** The GMSC is in charge of obtaining the ***MSRN*** **(*Mobile Station Roaming Number*)** from the HLR based on the MSISDN (Mobile Station ISDN number, the "directory number" of a MS) and routing the call to the correct visited MSC.
- SMS-G—This term is used to collectively describe the two ***Short Message Services Gateways*** described in the GSM recommendations. The SMS-GMSC (Short Message Service Gateway Mobile Switching Centre) is for mobile terminating short message, and the SMS-IWMSC (Short Message Service Inter-Working Mobile Switching Centre) for mobile originating short messages and transmitting to SMSC. The SMS-GMSC role is similar to that of the GMSC, whereas the SMS-IWMSC provides a fixed access point to the Short Message Service Centre.

6.1.4 Standards of CDMA

The technical standards of CDMA cellular mobile communication system are issued by ***American National Standard Institute*** **(*ANSI*)**. ***Telecommunications Industry Association*** **(*TIA*)**, a subordinate department of ANSI, mainly takes charge of developing Interim Standards (IS) serial standards such as IS-95、IS-41 and so on. These IS standards all have time limitation with 5 years' period of validity initially and now 3 years.

IS-95A is a technical standard of narrow CDMA (N-CDMA). The developing version of IS-

95A is IS-95B which can satisfy the requirement of higher bit rate traffics. The max bit rate that IS-95B can supply is as high as 115Kbps in theory, but only 64Kbps actually.IS-95A and IS-95B form IS-95 which is technical standards of the second generation CDMA cellular communication system. CDMA One is the appellative of all kinds of CDMA products with IS-95 as the core technical standard.

CDMA 2000, the 3G CDMA mobile communication standard instituted by TIA, is one of the three accredited standards **(*CDMA2000, W-CDMA, TD-SCDMA*)** relating to the 3G cellular mobile communication and a technique system scheme evolving from IS-95 to the 3G. It's a broadband CDMA technique. Its max ***indoor data rate*** is not less than 2 Mbps, 384Kbps on walking and over 144kbps on running automobile.

CDMA2000 1X is the first stage of CDMA 2000 and its data rate is higher than IS-95, but lower than 2Mb/s. It can support data transmission at the rate of 308Kb/s as well as mobile IP traffic when ***packet switching*** is adopted in the network.

6.1.5 Basic Parameters of CDMA Communication System

CDMA mobile communication system adopts ***CDMA/FDD*** access method, which means that the permitted channel bandwidth is firstly divided into many sub-channels, and then each sub-channel can support many links. Thus when a CDMA subscriber needs to start a communication process, he can be assigned an exclusive link through different PN sequence for his communication. The detailed CDMA sub-channels and corresponding frequency and the basic parameters of CDMA in 800MHz are respectively listed in Table 6-2 and Table 6-3. The minimum channel interval (central frequency difference between two carrier frequencies) is 1.23MHz.

Table 6-2 Basic parameters of CDMA Channels

	800MHz		**1800MHz**	
	Channel number	**Central frequency (MHz)**	**Channel number**	**Central frequency (MHz)**
MS	$1 \leqslant N \leqslant 777$	$0.03 \times N + 825.0$	$1 \leqslant N \leqslant 1199$	$0.050 \times N + 1850.00$
	$1013 \leqslant N \leqslant 1023$	$0.03 \times (N-102) + 825.0$		
BS	$1 \leqslant N \leqslant 777$	$0.03 \times N + 870.0$	$1 \leqslant N \leqslant 1199$	$0.050 \times N + 1930.00$
	$1013 \leqslant N \leqslant 1023$	$0.03 \times (N-1023) + 870.0$		

Table 6-3 Basic parameters of CDMA in 800MHz

	Parameters
Channel frequency	Down-chain: 869 ~ 894MHz(BS transmitting, MS receiving) Up-chain: 824 ~ 849MHz（BS receiving, MS transmitting）
Duplex mode	frequency division duplexing (FDD)
Transmission bandwidth	25MHz
Neighbor channels interval	1.25MHz
Access mode	CDMA/FDD
Modulation mode	Down: QPSK ; Up: OQPSK;
Channel coding	Down: convolutional code（r =1/2, k = 9） Up: convolutional code（r =1/3, k = 9）
Voice coding	CELP or QCELP coding algorithm
Data rate	9.6, 4.8, 2.4, 1.2 Kbps

The cycle length of pseudo random sequence in CDMA is 32768 (2^{15}) code chips, with 64 code chips merge to a modulation code pattern. It can hold 512 base stations in a CDMA cellular system with 1.25MHz bandwidth.

CELP (*Code Excited Linear Prediction*) coding algorithm is adopted in CDMA for voice code, which is also called QCELP (Qualcomm Code Excited Linear Prediction). By the CELP, audio signals can be transmitted with different rate of 9.6Kbps, 4.8 Kbps, 2.4 Kbps, 1.2 Kbps according to its different input.

CDMA cellular system utilizes timing mark of ***Global Position System* (*GPS*)**. Each base station is deployed with a GPS receiver for its ***Universal Timing*** reference.

6.1.6 Accessing CDMA Communication System

Accessing CDMA communication system is defined as the entire process from the MS's turning-on to its successful register in CDMA system. The state of this process is usually called initialization. As the first action of the handset during this period, it judges which mobile communication system it will work in, GSM or CDMA? This is determined by the relevant information stored in the SIM card (for GSM/GPRS system) or UIM card (for CDMA system) of the MS. If the latter one is chosen, the mobile station then starts to detect the ceaseless *pilot and synchronization signals* from surrounding CDMA base stations at once.

Although the PN sequences in pilot signal of each base station are of the same structure, they all have different offsets with each other. As long as the mobile station changes the offset of its local PN sequence, it can immediately detect the base stations around. By means of comparing intensities of these pilot signals, the MS makes choice of the present base station it should belong to at once.

Now the mobile station sends its request for accessing CDMA network to the selected BS through return channel. After receiving this request, the BS then transmits an ***authenticating parameter*** and encryption parameters group from the Visitor Location Registry of Mobile Switching Center (MSC/VLR) to the MS through forward channel.

According to a preset algorithm, then the MS calculates a response parameter by using parameters from the BS and the subscriber's identifying data for authentication stored in UIM card, and then submits the response parameter to the MSC/VLR through the BS. After comparison between the receiving response parameter and that in encryption parameter group in the MSC/VLR, the MS then will be regarded as a legal terminal and can immediately enjoy all corresponding CDMA services only when the two parameters are completely the same, otherwise, the MS will be regarded as a lawless one and denied the access to CDMA network.

6.1.7 Short Message

Existing as a subset of the larger context of second- and third-generation wireless technologies, ***Short Message Service* (*SMS*) contains *Cell Broadcast Service* (*CBS*)** and ***Point to Point Service* (*PPS*)**. It is the only one in all services of any mobile communication system that need not establish an end-to-end traffic link between mobile stations. In other word, even though a mobile station is in the process of communication or shut down, it is capable for the MS to receive messages to it.

Though a short message service is an asymmetric process, it's still a complete communication process for a mobile station to transmit or receive a short message, and, each message is independent on another.

Figure 6-3 shows the various components that make up the SMS cellular network. Notice that gateways provide mobile devices access to the Internet across TCP/IP lines.

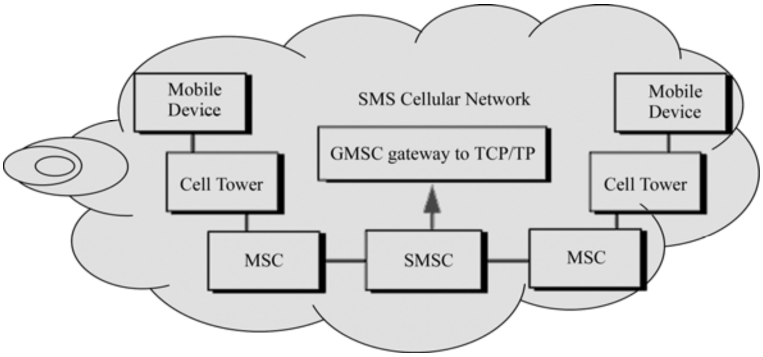


Figure 6-3 the route of a SM sent to its destination

A base station is the cellular relay station (or cell tower) that a cell phone talks to when initiating or receiving a wireless call. The base station's primary responsibility is to transmit voice and data traffic between mobile devices and a mobile switching center. All transmissions are managed by the base station, which acts as a kind of clearinghouse for wireless communications.

A Mobile Switching Center (MSC) is the electronic field office of a cellular carrier, a computer-controlled switch for managing automated network operations. An MSC automatically coordinates and controls call setup and routing between mobile phones in a given service area. MSCs are connected to base stations by T1 landlines or microwave channels, and by landlines to the Public Service Telephone Network (PSTN).

MSCs maintain individual subscriber records, current status of subscribers, and information on call routing and billing in two subscriber databases called the Home Location Register (HLR) and the Visitor Location Register (VLR). The HLR contains subscriber profiles, while the VLR provides information relevant to roamers.

SMS makes use of a Short Message Service Center (SMSC), which acts as a store-and-forward system for relaying short messages. Similar to an MSC, the SMSC guarantees delivery of text messages by the network. Messages are stored in the network until the destination cell phone becomes available, so a user can receive or transmit an SMS message at any time, whether a voice call is in progress or not.

SMSCs communicate with TCP/IP networks via a Gateway Mobile Switching Center (GMSC). A GMSC is an MSC capable of receiving short messages from an SMSC. The GMSC interrogates the Home Location Register (HLR) for subscriber routing information and delivers the short message to the home MSC or roaming MSC of the destination mobile unit.

Technical words and phrases

cellular ['seljulə] *adj.* 蜂窝

congestion [kən'dʒestʃən] *n.* 堵塞，阻塞（常发生于负载超过数据通路容量的时候）
arbitrarily ['ɑ:bitrerili] *ad.* 任意地，武断地，专横地
antenna [æn'tenə] *n.* 天线
boundary ['baundəri] *n.* 边界，界限，范围
handoff ['hændɔf] *n.* 切换
simultaneous [siml'teinjəs] *a.* 同时的，同时发生的
common air interface (CAI) 公共空中接口
forward voice channels (FVC) 前向语音信道
reverse voice channels (RVC) 后向语音信道
forward control channels (FCC) 前向控制信道
reverse control channels (RCC) 后向控制信道
roaming【计】 移动，移象，漫游
HLR——Home Location Register 本地位置寄存器
MIN （Mobile Identification Number）移动识别码
ESN （Electronic Serial Number）电子序列号
Global System for Mobile Communications (GSM) 全球移动通信系统（全球通）
popular ['pɔpjulə] *adj.* 通俗的，流行的，受欢迎的
throughout [θru:'aut] *prep.* 遍及，贯穿
protocol ['prəutəkəl] *n.* 草案，协议
Personal Communication Services (PCS) 个人通信服务
auctioned ['ɔ:kʃəned] *n. vt.* 拍卖
architecture ['ɑ:kitektʃə] *n.* 体系结构
emission [i'miʃən] *n.* 散发，发射，喷射
velocity [vi'lɔsiti] *n.* 速度，速率
district ['distrikt] *n.* 区域，地方，管区，行政区
subsystem ['sʌb'sistəm] *n.* 次要系统，子系统
entity ['entiti] *n.* 实体
MS——Mobile Station 移动台，移动终端
equipment [i'kwipmənt] *n.* 装备，设备
normal ['nɔ:ml] *adj.* 正常的，正规的，标准的
hand-held 手持式的
BTS——Base Transceiver Station 基站
device [di'vais] *n.* 装置，设备
BSC——Base Station Controller 基站控制器
release [ri'li:s] *n.* 释放
handover ['hændəuvə] *n.* 移交
BSS——Base Station Subsystem 基站子系统
MSC——Mobile Switching Centre 移动交换中心

VLR——Visitor Location Register　访问位置寄存器
permanent　['pə:mənənt]　*adj.*　永久的，持久的
temporary　['tempərəri]　*adj.*　暂时的，临时的，临时性
AuC——Authentication Centre　鉴权中心
database　['deitəbeis]　*n.*　数据库，资料库
semi-permanent　['semi-'pə:mənənt]　*adj.*　非永久（性）的，暂时的
EIR——Equipment Identity Register　设备身份寄存器
IMEI (International Mobile Subscriber Identity)　国际移动用户识别号
interrogate　[in'terəgeit]　*vt.*　审问，询问
GMSC——Gateway Mobile Switching Centre　移动交换中心网关
Mobile Station Roaming Number (MSRN)　移动台（终端）漫游号
Mobile Station Integrated Services Digital Network number
(Mobile Station ISDN number（MSISDN）　移动台的综合数字服务网（ISDN）号
route　[ru:t]　*v.*　路由
SMS-G——Short Message Services Gateways　短消息服务网关
recommendation　[rekəmen'deiʃən]　*n.*　推荐，建议
SMS-GMSC——Short Message Service Gateway Mobile Switching Centre
短消息服务-移动交换中心网关
SMS-IWMSC ——Short Message Service Inter-Working Mobile Switching Centre
短消息服务—内部移动交换中心
recurring　[ri'kə:riŋ]　*adj.*　经常性的，反复的
constitute　[kən'stitjut]　*vt.*　组成
bulletin　['bulitin]　*n.*　公告，报告；新闻简报
paging　[peidʃiŋ]　*n.*　寻呼
convey　[kən'vei]　*vt.*　搬运，传达，转让
burst　[bə:st]　*n.*　脉冲
session　['seʃən]　*n.*　会话
throughput　['θru:put]　*n.*　吞吐量；通过量
initialization　[iniʃəlai'zeiʃən]　*n.*　设定初值，初始化
intensity　[in'tensiti]　*n.*　强烈，剧烈，强度；亮度
authenticate　[ɔ:'θentikeit]　*v.*　鉴别，证明……是真的
encryption　[in'kripʃən] [计]　加密术，密码术
Visitor Location Registry of Mobile Switching Center (MSC/VLR)
移动交换中心的访问位置寄存器
preset　['pri:'set]　*vt.*　预先安置；预先调试
subscriber　[səb'skraibə]　*n.*　订户，签署者
submit　[səb'mit]　*vt.*　提交，递交
American National Standard Institute (ANSI)　美国国家标准协会

Telecommunications Industry Association (TIA)　（美国）电讯工业协会
subordinate　[sə'bɔ:dnit]　*adj.*　次要的，从属的，下级的
Interim Standards (IS)　临时标准，过渡标准
N-CDMA　窄带码分多址
traffic　['træfik]　*n.*　通信流量
appellative　[ə'pelətiv]　*adj.*　*n.*　通称名词，普通名词，称呼
accredited　[ə'kreditid]　*adj.*　可接受的，可信任的，公认的，质量合格的
evolve　[i'vɔlv]　*v.*　（使）发展，（使）进展，（使）进化
packet switching　分组交换
exclusive　[iks'klu:siv]　*adj.*　独占的，唯一的
code chip　码片
(Qualcomm) Code Excited Linear Prediction[(Q)CELP]　码激励线性预测编码
Global Position System (GPS)　全球定位系统
universal　[ju:ni'və:səl]　*adj.*　普遍的，全体的，通用的
integrated　['intəgreitid]　*adj.*　集成的
Short Message Service (SMS)　短消息业务
Cell Broadcast Service (CBS)　小区广播业务
Point to Point Service (PPS)　点到点业务
end-to-end　端到端
asymmetric　[æsi'metrik]　*adj.*　不均匀的，不对称的
Short Message Service Centre (SMSC)　短消息服务（业务）中心
infrastructure　['infrəstrʌktʃə]　*n.*　基础设施
Short Message Entity(SME)　短消息实体
popedom　['pəupdəm]　*n.*　权限；辖区
impart　[im'pɑ:t]　*vt.*　给予（尤指抽象事物），传授，告知，透露
overstep　['əuvə'step]　*vt.*　踏过，逾越，超出……的限度
coverage　['kʌvəridʒ]　*n.*　覆盖的范围
entrance switching　入口交换机
exit switching　出口交换机

6.2　Reading Materials

1．UIM Card

UIM card is a smart microprocessor with integrated CPU, ROM, RAM, EPROM/E^{2}PROM and serial communication unit on it, while abbreviation UIM means *User Identity Model.* Each UIM card preserves relevant information of its user and others, which can help the MS's operation through the access control to all types of data, i.e., the subscriber's identity authentication calculation. Generally, information in a UIM card can be approximately divided into the following three kinds.

① Information for user identification and authentication: such as the *International Mobile Subscriber Identification Number* (*IMSI*), the proprietary authentication parameters of CDMA system etc.

② Service information：information about traffics in CDMA system stored in HLR, such as state of short message.

③ Information relating to the MS's performance：majority are parameters, such as preferred system and channel and so on.

Besides, subscriber can save his own information in UIM card, such as DN.

2．China Mobile Launches New Bank Service

Beijing Liandong Youshi, an affiliate of China Mobile which is responsible for the company's mobile phone banking service, has announced that China Mobile has reached an agreement with 19 domestic banks to jointly promote a mobile phone SMS-based banking service.

With this new service, which is called "Yin Xin Tong", mobile phone users will be able to obtain financal information through their mobile phones and wireless POS by sending text messages or logging in through GPRS and WAP.

A representative from China Mobile has told local media that some of the 19 banks include Industrial and Commercial Bank of China, China Construction Bank, China Merchants Bank, Industrial Bank and Huaxia Bank. According to their agreements, the 19 banks will buy a mobile phone short message and group message sending terminal from China Mobile to provide financal service to their clients.

At present, the service can be used to remind users about card use, notify them about account transfers and loan payments, and make inquiries on bank accounts, bank transaction, stock and foreign exchange information.

UMPay says that because the service is being tested, it is free for the time being. But a detailed charging system will come out soon.

UMPay is a joint venture between China Mobile and China UnionPay. Each holds 50% stake in the company.

(Source: Chinatechnews.com)

3．RAKE Reception Technique

Fading and distortion are inevitable for signals transmitted in mobile communication channel because of the unideal characteristics of channel, and then make bad influence on system performance. The basic theory of RAKE reception technique is: by means of collecting all transmitted signals from multi-path and uniting them, the received signal power can obtain an effective enhancement, which can improve the output SNR and system performance.

Three, four RAKE receivers are respectively set in each MS, BS in CDMA system in order to receive the arriving branch signals of the same original signal from different transmitting routes, and then combine them to get an increscent output Signal-to-Noise ratio (SNR) after their respectively independent demodulation. Thus, in favor of RAKE reception technique, the disadvantageous factor

that signal transmitted in multi-path becomes an advantageous factor in CDMA cellular mobile communication system.

4．Low Probability of Intercept (LPI) in CDMA

It is necessary for eavesdropping somebody's conversation that the transmitted conversation information must be captured and then decoded. For CDMA system, signal with user's information must be spread spectrum modulated before sent to the common channel with extremely spread spectrum and very low power spectral density. Thus, it is too difficult to detect this kind of CDMA signal for the listener-in because its power spectral density is even lower than ambient noise, and it's almost impossible for him to distinguish the signal from background noise.

Moreover, even if the signal is captured, what the listener-in can get is only broad band noise of insignificance from his demodulator because of his inaccurate local pseudo-random sequence. The probability of right PN sequence pattern necessary for spread spectrum demodulation is one of more than one thousand billion! What a low probability for listener-in to eavesdrop other's conversation through CDMA cellular mobile communication system!

5．Few Off-line Chances in CDMA

Because of the soft-switching technique, which means "connect-before-break", it's overcome for CDMA that the frequent occurrence of off-line.

In mobile communication system, base station (BS) is the guarantee for conversation. When subscribers during conversation moving close to the edge of the BS's demesne, the BS should maintain the communication through its active switching, otherwise the process will be interrupted.

During switching period, BS's dominion should shift from "demesne of the local BS (such as A) " to "demesne of A and its neighboring BS (B)" to "demesne of B", and signal from the moving mobile station will automatically switch to a relatively idle neighboring BS. Thus, only when it is approved that the moving MS has moved to the "demesne of B", can the connection between the moving MS and its original BS (A) be switched off. So it is not easy for a CDMA conversation to be interrupted by off-line.

Unlike the above soft-switching technique in CDMA, hard-switching method adopted in GSM system executes switching process as "break-before-connect", which happen to be the reversed processing sequence with that of CDMA. That is to say, when a GSM subscriber during conversation moving close to edge of the BS's demesne, the local BS (A) cut off connection with the MS at first, then the MS set new connection with the neighboring BS (B) once again. Thus, communication during this switching will have to de interrupted, that consequently leads to a high off-line ratio.

6.3　Exercises

1．Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）国际移动用户识别码

（2）归属位置寄存器
（3）访问位置寄存器
（4）移动交换中心
（5）基站（子）系统
（6）全球移动通信系统
（7）个人通信业务
（8）交换子系统
（9）手持蜂窝电话
（10）基站控制器
（11）鉴权中心
（12）移动交换中心网关
（13）移动台漫游号码
（14）物理信道
（15）逻辑信道
（16）码激励线性预测编码
（17）全球定位系统

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）The full name of GSM is （　　）, which is one of the most popular mobile communication mode throughout the world. The term GSM usually means the GSM（　　）and protocols in the frequency spectrum around 900MHz. The original and most widely used GSM frequency implementation is also becoming known as （　　）, and（　　）is also known as GSM1800, which has different physical frequencies but the same（　　）and architecture with GSM900.

（2）The emitting power of each GSM base station is 500 W per carrier wave, while each mobile station has five emitting power to chose, namely （　　）W, 2W, 5W, 8W and （　　）W. The coverage radius of a small district in GSM 900 is （　　）km in maximum and （　　）m in minimum. There are three basic elements in a GSM mobile communication system, the（　　）subsystem, the（　　）subsystem and other networks.

（3）Mobile Station can be briefly expressed as （　　）, which is the physical equipment used by subscriber. The entire expression of BTS is （　　）, comprising the radio transmission and（　　）devices for purpose of processing signal related to the（　　）interface.

（4）The VLR in GSM system contains all the（　　）data, both（　　）and temporary, which are necessary to control a MS in the MSC’s coverage area. VLR is commonly recognized as an integral part of （　　）, rather than a separate entity. Similarly, the HLR is also a （　　）used to store（　　）and semi-permanent subscriber data, which always knows the（　　）of a MS.

（5）Access to CDMA communication system is defined as the entire process from the MS’s （　　）to its successful（　　）in CDMA system. The state of this process is usually

called （　　）. As the first action of the handset during this period, it（　　）which mobile communication system it will work in, GSM or CDMA?

（6）The technical standards of CDMA cellular mobile communication system are issued by （　　）. TIA mainly takes charge of developing Interim Standards serial such as （　　）、IS-41 and so on. These IS standards all have time（　　）with 5 years' period of （　　）initially and now（　　）years.

（7）IS-95A is a technical standard of （　　）. The developing version of IS-95A is（　　）which can satisfy the requirement of higher bit rate traffics, with the max bit rate of 115Kbps in theory but only （　　）Kbps actually.

（8）The three accredited standards of 3G cellular mobile communication is CDMS 2000, （　　）and （　　）. CDMA 2000 is a broadband CDMA technique evolved from IS-95, with the max indoor data rate no less than （　　）Mbps, 384Kbps on walking and over （　　）Kbps on running automobile.

（9）CDMA mobile communication system adopts（　　）access method. When a CDMA subscriber needs to start a communication process, he can be assigned an exclusive （　　）through different（　　）sequence for his communication. The minimum channel interval (central frequency difference between two carrier frequencies) is （　　）MHz.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（　　）The emitting power of each GSM base station is 500 W per carrier wave.

（2）（　　）BSS means Base Station Subsystem, which consists of a BSC and a BTS.

（3）（　　）BSC is the abbreviation of Base Station Controller, which manages the physical interface, mainly through the allocation, release and handover of physical channels.

（4）（　　）AuC means Authentication Centre, containing the subscriber authentication keys and the algorithm required to calculate the authentication parameters to be transferred to the HLR.

（5）（　　）MSC means Mobile Switching Centre, which is basically an ISDN-switch, coordinating and setting up calls to and from MSs.

（6）（　　）EIR means Equipment Identity Register, a database contains information of the BS and its capabilities.

（7）（　　）IMEI is the abbreviation of International Mobile Subscriber Identity, which is used to interrogate the EIR.

（8）（　　）HLR is a database used to store permanent subscriber data, which always knows the location area of the MS (assuming the MS is in a coverage area), and this data is used to locate an MS in the event of a MS terminating call set-up.

（9）（　　）VLR means Visitor Location Register, which contains all the subscriber data, both permanent and temporary, which are a necessary entity independent with MSC.

（10）（　　）After receiving the request of accessing to CDMA network from MS, the BS transmits an authenticating parameter and encryption parameters group from the Visitor Location Registry of Mobile Switching Center to the MS through forward channel.

（11）（　　）The technical standards of CDMA cellular mobile communication system are issued by Telecommunications Industry Association.
（12）（　　）CDMA2000 1X is the alternative of CDMA 2000 with the data rate higher than IS-95 but lower than 2Mbps.
（13）（　　）CDMA cellular system utilizes timing mark of GPS, deployed in each BS with a GPS receiver for its Universal Timing reference
（14）（　　）Generally, information in a UIM card can be approximately divided into three kinds: Information for user identification and authentication; Information relating to the MS's performance and subscriber's private information.
（15）（　　）When a mobile station is shut down, it could not receive messages to it.
（16）（　　）Because a short message service is an asymmetric process, it's not a complete communication process.

6.4　课文参考译文量　2G 蜂窝通信系统

6.4.1　通信系统中的蜂窝技术

蜂窝概念在解决带宽瓶颈和用户容量这两个问题上是一项重大的突破。蜂窝概念在有限频带范围内提供了非常高的容量，而不需要任何技术上的转变。蜂窝这个概念属于系统级别，它取代了原先单一的、高功率的发射机（大蜂窝）以大量的小功率发射机（小蜂窝），每个小功率发射机只覆盖一小片服务区域。每个基站分配了一部分信道，这样剩下的其余可用信道分配给了相邻的少数几个基站。因为相邻信道分配了不同的信道，所以基站（包括从属于这些基站的用户）之间的干扰就可以降低到最低。

由于业务需求的增加（如市场需要更多的信道），基站的数量也需要增加（随之而来的是发射功率的降低以避免干扰），这样也就提供了额外的无线用户容量，但是却不需要增加无线频谱。这个基本原理是所有现代无线通讯系统的理论基础，因为，通过在整个覆盖区域内重复利用信道，固定数量的信道就可以满足任意数量用户的需要。另外，蜂窝概念让一个国家或一片大陆上所有用户设备以相同的制式来生产，这样移动设备就可以在该地区使用。

蜂窝无线系统依赖区域内信道的智能分配和重复利用。每个蜂窝基站分配了一组无线信道，在一个称为“小区”的地理区域内使用。任意一个小区内分配的信道与相邻小区的信道都是不同的。基站天线用来覆盖特定的一个小区。通过将基站的覆盖范围限制在小区的边界以内，同一组信道就可以反复地为不同的小区使用，只要这些小区之间的距离足够大，这样互相之间的干扰就可以降低到允许的范围。整个蜂窝基站系统内部无线信道群选取和分配的设计过程称为频率复用或频率规划。

只要用户处在蜂窝电话系统的无线覆盖范围之内，该系统就可以为用户提供 PSTN 的无线接入。蜂窝无线系统可以提供媲美于固定电话的高质量服务。通过将基站发射机的覆盖范围限制在称为“小区”的一小块地理区域之内，同一组无线信道可以被远方的另一个基站重复使用，这样蜂窝系统就可以实现很高的容量。当用户进入一个新的小区时，一种称为“切换”的交换技术使得通话不会受到中断。

移动台由收发机，天线和控制电路等组成，通常安装在车辆或手持移动终端上。基站由几个

发射机和接收机组成，这些发射机和接收机用于处理全双工通信，发射天线和接受天线则安装在基站的高塔上。基站在小区内所有的移动用户之间起着桥梁的作用，通过通信电缆或微波链路将移动呼叫路由到基站控制器（MSC）。基站控制器一般可以容纳 100 000 个蜂窝用户，以及同时处理 5 000 个呼叫，当然也可以实现所有的计费和系统维护功能。

基站和移动台之间的通信信道由标准的公共空中接口（CAI）来定义，该接口包括四种不同的信道。用于传输由基站到移动台的语音通信信道称为前向控制信道（FVC），用于传输由移动台到基站的语音通信信道称为反向语音信道（RVC）。发起移动呼叫的信道是前向控制信道（FCC）和反向控制信道（RCC）。

所有的蜂窝系统均提供“漫游”服务。漫游让用户的手机可以在该用户注册的服务区域之外继续使用。当移动台进入城市或某个地区时，如果这个区域不是其注册的服务区域，该移动台就在新的服务区域内注册为“漫游者”。这个过程通过前向控制信道（FCC）来完成。每数分钟，基站控制器（MSC）就给系统中的所有前向控制信道广播一条全局指令，要求所有尚未注册的移动台将其移动用户识别码（MIN）和电子序列号（ESN）通过反向控制信道报告给基站控制器。系统中未注册的移动台在收到注册请求后，将周期性地汇报其用户信息，然后基站控制器使用用户识别码（MIN）/电子序列号（ESN）数据从归属用户寄存器（HLR）那里查询“漫游”移动台计费状态。一旦注册后，漫游移动台将可以在该区域发起呼叫或接听呼叫，而计费信息则自动地路由到该用户的归属业务供应商那里。

6.4.2 GSM 简介

全球移动通信系统（GSM）是目前使用最为广泛的两个移动通信系统之一，通常所说的 GSM 指的是 900MHz 频段下的 GSM 标准及协议。实际上 GSM 还有一个 DSC1800 频段，它使用与 900MHz 频段相同的协议和不同的空中接口。此外，在美国，还有一个付费的用于个人通信服务（PCS）的 GSM1900MHz 频段。总之，GSM 系统最初也是目前使用最多的频段是 900MHz 频段，一般用 GSM900 表示，类似地，DCS1800 也表示为 GSM1800。两者虽然使用频率不同，但具有相同的系统结构并使用同样的协议。

GSM 系统基本参数如下表 6-1 所示。

译表 6-1　GSM 基本参数

	GSM900	GSM1800	GSM1900
发射频率（MHz）	890～915	1710～1785	1850～1910
接收频率（MHz）	935～960	1805～1880	1930～1990
复用方式	FDD		
频道间隙（kHz）	200		
手机发送功率（最大/平均）(mW)	1000 / 125		
功率控制（手机，基站子系统）	Yes		
语音编码/ 速率 (Kbps)	RPE-LTP / 13		
信道速率 (Kbps)	270.833		
信道编码	1/2 码速 卷积码		
帧周期 (ms)	4.615	374	299
调制方式	GMSK		
蜂窝半径 (km)	<35	<4	<4
移动速率 (km/hour)	250	125	125

基站每个载频的发射功率为 500W，移动台（手机）发射功率有 5 个等级可选：0.8W、2W、5W、8W 和 20W。一个 GSM900 的蜂窝小区覆盖范围可由最小半径 500m 到最大半径 35km。

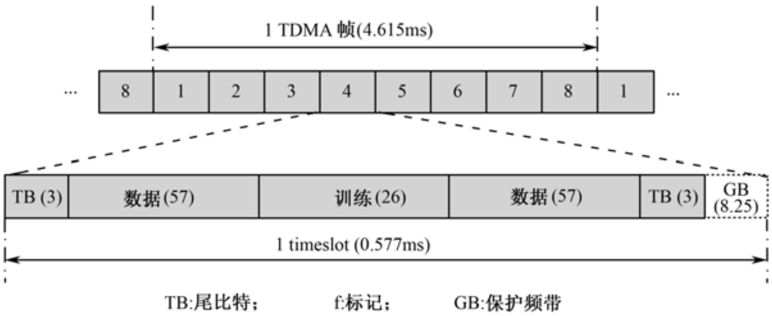
6.4.3　GSM 系统结构

图 6-1 所示为 GSM 系统时分多址接入形式及其按 22.8 Kbps 速率进行信道编码、交织后的常规 Burst 结构，它具有 13Kbps 的全语音编码速率和最高 9.6Kbps 的数据传输速率。GSM 还为该 TDMA 结构定义了半速率服务，提供 11Kbps 的总速率和 4.8Kbps 的数据速率。

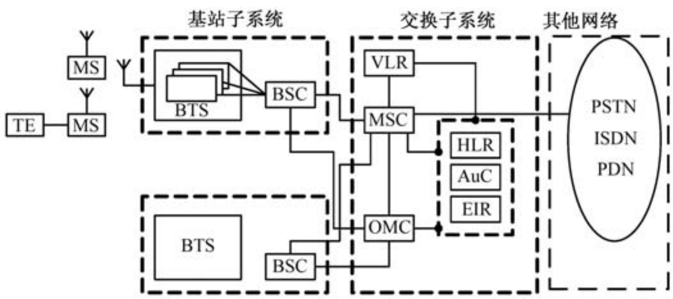
GSM 系统包含三个基本组成部分：基站子系统、交换子系统和其他网络，如图 6-2 所示。

图中 GSM 主要功能实体简介如下。

- 移动台（MS）—— GSM 用户使用的终端设备，一般为手机。
- 基站收发台（BTS）—— 含无线发送和接收设备，并实现信号与空中接口的连接。



译图 6-1　GSM 的时分复用结构



译图 6-2　GSM 系统构架

- 基站控制器（BSC）——实现无线接口管理，主要包括无线信道的分配、释放和转移。
- 基站子系统（BSS）——基站控制器以及多个基站的集合。
- 移动交换中心（MSC）——主要由 ISDN 交换机构成，调整并建立来自、发往移动台的呼叫处理。
- 访问位置寄存器（VLR）——存储用户的所有数据，包括 MSC 辖区内每个移动台所必需的临时性数据和永久性数据。VLR 常常被看作为移动交换中心 MSC 的一个部分而非一个单独的功能实体。
- 鉴权中心（AuC）——含有用户鉴定密码以及利用送到 HLR 的鉴权参数计算鉴权数值算法的数据库。
- 归属位置寄存器（HLR）——用于存储暂时和永久用户数据的数据库，使系统交换

中心了解用户所在位置，并由此判断某通信用户是否属于漫游状态。

- 设备身份寄存器（EIR）——用于存储用户终端设备相关信息的存储器，国际移动用户序列号（IMEI）就用于 EIR 查询。
- 移动交换中心网关（GMSC）——也称关口局，主要（起汇接功能），根据 HLR 中的相关移动台 ISDN 号（移动台的“目录”数据）获得该移动台漫游数据（MSRN），将与其相关的呼叫路由至移动台当前所在位置的交换中心 MSC（保证正确的通信连接）。
- 短消息关口（SMS-G）—— 该缩写词指 GSM 建议标准中的两个短消息服务网关：SMS-GMSC 和 SMS-IWMSC。其中，短消息服务-移动交换中心网关（SMS-GMSC）接收发自 SMSC 短消息；短消息服务-内部移动交换中心（SMS-IWMSC）则接收来自网络的短消息，并将此消息送到相应 SMSC。SMS-GMSC 类似于 GMSC，而 SMS-IWMSC 则为短消息服务中心提供固定接入点。

6.4.4 CDMA 标准

CDMA 蜂窝移动通信系统的技术标准由美国国家标准协会（ANSI）发布，ANSI 旗下的电讯工业协会（TIA）主要负责其中的过渡性标准（IS）系列，如 IS-95、IS-41 等。起初这类过渡性标准的有效时限是 5 年，现在已改为 3 年。

IS-95A 是窄带 CDMA 标准，其后续版本 IS-95B 可支持更高的传输速率，尽管 95B 的实际最高速率只有 64Kbps，其理论最高比特率为 115Kbps。IS-95A 和 IS-95B 构成了第二代 CDMA 蜂窝移动通信系统的技术标准。所有以 IS-95 为核心标准的产品均统称为 CDMA One.

CDMA2000 是 TIA 发布的第三代（3G）CDMA 移动通信标准，是目前公认的 3 个 3G 蜂窝移动通信标准（CDMA2000、W-CDMA、TD-SCDMA）之一，也是由窄带 IS-95CDMA 向宽带 3G 移动通信系统过渡的技术下体系规范。CDMA2000 的最大数据传输速率分别达到室内 2Mbps、步行 384Kbps 和行车 144Kb/s。

CDMA20001X 是 CDMA2000 的第一个过渡性标准，其数据速率可达 308Kb/s，高于 IS-95 但低于 2Mbps，并在网络中采用分组交换方式以支持移动 IP 数据传输。

6.4.5 CDMA 系统基本参数

CDMA 移动通信系统采用码分多址/频分复用（CDMA/FDD）的信道多址与复用方式，首先将其规定的传输频段带宽按频分方式划分为多个子信道，再使每个子信道按码分多址技术接入多个链接，使每个 CDMA 用户在需要通信时即可对其分配唯一的伪随机（PN）序列，使其可在通信过程中独占一个链接。详细的 CDMA 子信道及其相应频率以及系统 800MHz 频段的基本参数分别如表 6-2 和 6-3 所列。最小信道间隔（两个载波的中心频率之差）为 1.23MHz。

译表 6-2 CDMA 信道的基本参数

	800MHz		1800MHz	
	信道数	中心频率（MHz）	信道数	中心频率（MHz）
移动台	$1 \leqslant N \leqslant 777$	$0.03 \times N + 825.0$	$1 \leqslant N \leqslant 1199$	$0.050 \times N + 1850.00$
	$1013 \leqslant N \leqslant 1023$	$0.03 \times (N-102) + 825.0$		
基站	$1 \leqslant N \leqslant 777$	$0.03 \times N + 870.0$	$1 \leqslant N \leqslant 1199$	$0.050 \times N + 1930.00$
	$1013 \leqslant N \leqslant 1023$	$0.03 \times (N-1023) + 870.0$		

表 6-3 800MHz CDMA 系统基本参数

	参　　数
信道频率	下行: 869～894MHz(基站发射, 移动台接受) 上行：824～849MHz（基站接受，移动台发射）
双工模式	频分双工（FDD）
带宽	25MHz
相邻信道间隔	1.25MHz
接入方式	CDMA/FDD
调制方式	下行：QPSK ；上行：OQPSK；
信道编码	下行：卷积编码（$r = 1/2$，$k = 9$） 上行：卷积编码 e（$r = 1/3$，$k = 9$）
语音编码	CELP or QCELP 语音编码
数据速率	9.6, 4.8, 2.4, 1.2 Kbps

CDMA 中的伪随机序列周期长度为 32768 即 2^{15} 个码片，每 64 个码片组成一个调制码型。因此，CDMA 蜂窝移动通信系统每 1.25MHz 带宽内可容纳 512 个基站。

CDMA 采用码激励线性编码（简称 CELP 或 QCELP），可根据输入语音信号的不同分别按 9.6Kbps、4.8 Kbps、2.4 Kbps 和 1.2 Kbps 几种速率编码传送。

CDMA 中采用全球定位系统（GPS）的时标，在其每个基站内都配置了 GPS 接收器为其提供格林尼治时间参照。

6.4.6 接入 CDMA 通信系统

一般将 CDMA 手机从开机到成功登录系统的整个过程称为接入 CDMA 系统，与此相应的手机状态则称为初始化状态。一旦开机，手机首先就根据其 SIM 卡（对 GSM 手机而言）或 UIM 卡（对 CDMA 手机而言）上的相关信息判断其所属通信系统是 GSM 还是 CDMA？若答案为后者，则手机立即开始搜索附近 CDMA 基站不断发出的导频信号和同步信号。

尽管每个 CDMA 基站发出的导频信号中的伪随机（PN）序列具有完全相同的结构，但它们彼此间的偏移却互不相同。手机一旦将其本地 PN 序列的偏移与基站调准，它即可立即搜索到其所在地附近的基站,并通过比较收到的导频信号强度，迅速选定它的归属基站。

此后，该手机通过后向信道向其选定的基站发出入网请求。接收到该请求后，基站通过前向信道向手机返回从移动交换中心的访问位置寄存器（MSC/VLR）获取的一个鉴权参数以及加密参数组。

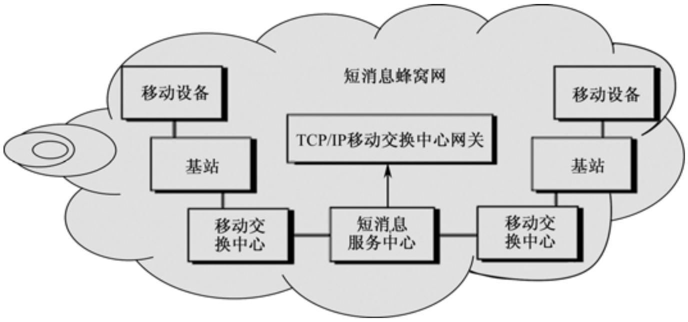
按照某预设的算法，手机利用收到的参数以及 UIM 卡上存储的相关用户身份识别数据，得出相应的响应结果，并将其通过归属的基站提交给 MSC/VLR，由它将计算结果与数据库中存储的相应加密参数组对比判断。只有当两者完全相同时，才认定该手机为合法用户并可立即得到所有相应的服务；否则，就认为该手机为非法用户，并拒绝其入网。

6.4.7 短消息

短消息业务是第二代、第三代无线通信相关技术的一个子集，包括小区广播业务（CBS）和点对点业务（PPS）两类。短消息是移动通信系统中唯一无需端到端的连接即可实现的业

务，即使一个手机正处于通话进程或已经关机，它仍然能够接收到发给它的短消息。尽管短消息服务是一个非对称的过程，其收、发短消息的过程依然可算是一个完整的通信过程，并不依赖于其他服务项目。

图 6-3 画出了短消息蜂窝网的组成，及其通过网关接入 ICP/IP 的链路。



译图 6-3　SMS 蜂窝网

基站是一个蜂窝中继站（或基站），当发出或者接收一个无线呼叫时，一部手机同这个基站通话。基站的主要职责是在移动设备和移动交换中心间传送语音和数据流量。所有的传输由基站来管理，基站作为无线通信的交换中心。

移动交换中心（MSC）是蜂窝载波的电磁场办公中心，由一台计算机控制的交换中心管理自动的网络操作。一个 MSC 自动协调和控制在特定服务区域内的移动手机之间的呼叫和连接。MSC 经过 T1 线和微波信道连接到基站，然后经过线缆连接到公众业务电话网（PSTN）。

MSC 在两个叫做归属位置寄存器（HLR）和拜访位置寄存器（VLR）的用户数据库中保存个人用户记录、当前用户的状态和存储呼叫路由与计费信息。HLR 包含用户信息，而 VLR 提供与漫游相关的信息。

SMS 充分利用作为一个用于中继短消息的存储转发系统的短消息业务中心（SMSC）。与 MSC 很类似，SMSC 确保网络上传送文本消息。在指定的手机可用之前，消息一直存储在网络上，所以用户可以在任何时候接收或发送一条 SMS 消息，而同样情况下，语音呼叫就不可以。

通过一个网关移动交换中心（GMSC），网关移动交换中心与 TCP/IP 网络通信。GMSC 是一台具有从短消息业务中心（SMSC）上接受短消息的能力的 MSC。GMSC 包含用来查询用户路由信息的归属位置寄存器（HLR）和投递短消息给指定移动单元的归属 MSC 或漫游 MSC。

Unit7　PBX system and Signaling System

7.1　Text

7.1.1　PBX

A ***PBX*** is a telephony device that is used by most medium- and larger-sized companies. A PBX enables users or subscribers of the PBX to share a certain number of outside lines for making telephone calls. A PBX is a much less expensive solution than giving each user in a business a dedicated external telephone line. Telephone sets, in addition to fax machines, modems, and many other communication devices, can be connected to a PBX.

The PBX equipment is typically installed at a business's premises and connects calls between the telephones located and installed in the business site. A limited number of outside lines, also known as trunk lines, are typically available for making and receiving calls that are external to the business from an external source such as the ***PSTN***.

Internal business calls made to external telephone numbers by using a PBX are made by dialing 9 or 0 in some systems followed by the external number. An outgoing ***trunk line*** is automatically selected to complete the call. Conversely, the calls placed between users within the business do not ordinarily require special dialing digits or use of an external trunk line. This is because the internal calls are routed or switched by the PBX between telephones that are physically connected to the PBX.

In medium- and larger-sized businesses, the following PBX configurations are possible:

- A single PBX that supports the whole business.
- A grouping of two or more PBXs that are not networked or connected to each other.

7.1.2　PBX Systems

PBXs contain line cards that support various transmission protocols such as ***ISDN***, ***T1/E1***, ***HDSL***, and ***ADSL*** (Figure 7-1). PBXs also have features such as a ***POTS*** (plain old telephone service) pull-through which allows stations to have outside line access in the event of power failure.

7.1.3　IP PBX

An ***IP PBX*** is a Private Branch eXchange (PBX) that supports the IP protocol to connect phones by using an Ethernet or ***packet-switched*** LAN and sends its voice conversations in IP packets. A hybrid IP PBX supports the IP protocol for sending voice conversations in packets, but also connects traditional analog and digital ***circuit-switched Time Division Multiplex*** **(*TDM*)** telephones. An IP PBX is telephone switching equipment that resides in a private business instead of the telephone company.

One of the main advantages of an IP PBX is the fact that it employs converged data and voice networks. This means that Internet access, as well as VoIP communications and traditional telephone communications, are all possible using a single line to each user. This provides flexibility

as an enterprise grows, and can also reduce long-term operation and maintenance costs. Like a traditional PBX, an IP PBX is owned by the enterprise.

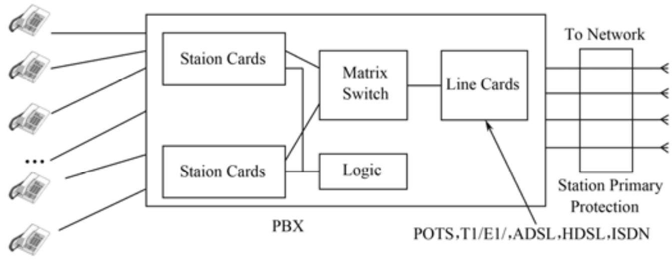


Figure 7-1 PBX Overview

IP PBXs are frequently easier to administer than legacy PBXs, because administrators can easily configure their IP PBX services by using an Internet browser or another IP-based utility. Plus, no additional wiring, cabling, or patch panels must be installed. With an IP PBX, moving an IP-based telephone is as simple as unplugging a telephone and plugging it in at a new location. Additionally, businesses that own an IP PBX do not have the additional infrastructure costs that are required to maintain and manage two separate circuit-switched and packet-switched networks.

7.1.4 Centrex

Centrex is a set of specialized business solutions (primarily, but not exclusively, for voice service) where the equipment providing the call control and service logic functions is owned and operated by the service provider and hence is located on the service provider's premises. Since Centrex frees the customer from the costs and responsibilities of major equipment ownership, Centrex can be thought of as an outsourcing solution.

Call control and service logic refer collectively to the functions needed to process a telephone call and offer telephone features. The following are examples of call control and service logic functions:

- recognizing that a party has gone off hook and that dial tone should be provided
- interpreting the dialed digits to determine where the call is to terminated
- determining whether the called party is available, busy, or has call forwarding, and then applying the appropriate treatment (e.g., ringing the phone, applying busy signal, applying a call waiting tone, delivering the call to voicemail, or forwarding the call to another party)
- recognizing when the called party answers the phone and when either party subsequently hangs up, and recording the appropriate information for billing

In traditional Centrex service (i.e., analog Centrex and ISDN Centrex), call control and service logic reside in a ***Class 5 switch*** located in the Central Office. The Class 5 switch is also responsibility for transporting and switching the electrical signals that carry the callers' speech or other information (e.g., faxes).

7.1.5 IP Centrex

In IP telephony, voice conversations can be digitized and packetized for transmission across the network. IP Centrex refers to a number of IP telephony solutions where Centrex service is

offered to a customer who transmits its voice calls to the network as packetized streams across a broadband access facility. IP Centrex builds on the traditional benefits of Centrex by combining them with the benefits of IP telephony. One of these IP telephony benefits is increased utilization of access capacity. In IP Centrex, a single broadband access facility is used to carry the packetized voice streams for many simultaneous calls. When calls are not active, more bandwidth is available for high speed data sessions over the ***LAN***.

7.1.6 Signaling System

There are two essential components to all telephone calls. The first, and most obvious, is the actual content—our voices, faxes, modem data, etc. The second is the information that instructs telephone exchanges to establish connections and route the "content" to an appropriate destination. ***Telephony signaling*** is concerned with the creation of standards for the latter to achieve the former. These standards are known as protocols. ***SS*7** or ***Signaling System Number* 7** is simply another set of protocols that describe a means of communication between telephone switches in public telephone networks. They have been created and controlled by various bodies around the world, which leads to some specific local variations, but the principal organization with responsibility for their administration is the ***International Telecommunications Union*** or ***ITU-T***.

Signaling System Number 7 (SS#7 or C7) is the protocol used by the telephone companies for ***interoffice signaling***. In the past, in-band signaling techniques were used on interoffice trunks. This method of signaling used the same physical path for both the call-control signaling and the actual connected call. This method of signaling is inefficient and is rapidly being replaced by out-of-band or common-channel signaling techniques.

To understand SS7 we must first understand something of the basic inefficiency of previous signaling methods utilized in the Public Switched Telephone Network (PSTN). Until relatively recently, all telephone connections were managed by a variety of techniques centered on "in band" signaling. A network utilizing common-channel signaling is actually two networks in one:

1. First there is the circuit-switched "user" network which actually carries the user voice and data traffic. It provides a physical path between the source and destination.

2. The second is the signaling network which carries the call control traffic. It is a packet-switched network using a common channel switching protocol.

The original common channel interoffice signaling protocols were based on Signaling System Number 6 (SS#6). Today SS#7 is being used in new installations worldwide. SS#7 is the defined interoffice signaling protocol for ISDN. It is also in common use today outside of the ISDN environment.

The primary function of SS#7 is to provide call control, remote network management, and maintenance capabilities for the inter- office telephone network. SS#7 performs these functions by exchanging control messages between SS#7 telephone exchanges (***signaling points*** or ***SPs***) and SS#7 signaling transfer points (STPs).

The switching offices (SPs) handle the SS#7 control network as well as the user circuit-switched network. Basically, the SS#7 control network tells the switching office which paths to

establish over the circuit-switched network. The STPs route SS#7 control packets across the signaling network. A switching office may or may not be an STP.

Signaling Links

SS7 messages are exchanged between network elements over 56 or 64 kilobit per second (Kbps) bidirectional channels called signaling links. Signaling occurs out-of-band on dedicated channels rather than in-band on voice channels. Compared to in-band signaling, out-of-band signaling provides:

- faster call setup times
- support for ***Intelligent Network*** **(*IN*)** services which require signaling to network elements without voice trunks (e.g., database systems)
- improved control over fraudulent network usage

Signaling Points

Each signaling point in the SS7 network is uniquely identified by a numeric point code. Point codes are carried in signaling messages exchanged between signaling points to identify the source and destination of each message. Each signaling point uses a routing table to select the appropriate signaling path for each message.

There are three kinds of signaling points in the SS7 network (Figure 7-2):

- ***SSP*** **(*Service Switching Point*)**
- ***STP*** **(*Signal Transfer Point*)**
- ***SCP*** **(*Service Control Point*)**

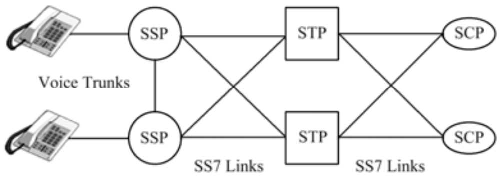


Figure 7-2　SS7 network

SS7 Protocol Stack

The hardware and software functions of the SS7 protocol are divided into functional abstractions called "levels". These levels map loosely to the ***Open Systems Interconnect*** **(*OSI*)** **7-*layer model*** defined by the ***International Standards Organization*** **(*ISO*)**, described concretely in Figure 7-3.

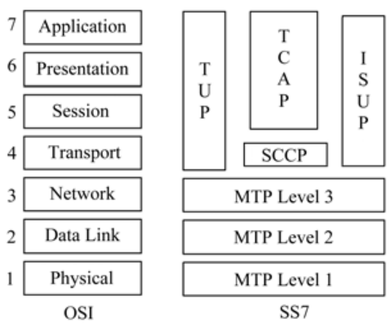


Figure 7-3　The OSI Reference Model and the SS7 Protocol Stack

Message Transfer Part

The ***Message Transfer Part*** (***MTP***) is divided into three levels. The lowest level, MTP Level 1, is equivalent to the OSI Physical Layer. MTP Level 1 defines the physical, electrical, and functional characteristics of the digital signaling link. Physical interfaces defined include E-1 (2048 Kbps; 32 64 Kbps channels), DS-1 (1544 Kbps; 24 64 Kbps channels), V.35 (64 Kbps), DS-0 (64 Kbps), and DS-0A (56 Kbps).

MTP Level 2 ensures accurate end-to-end transmission of a message across a signaling link. Level 2 implements flow control, message sequence validation, and error checking. When an error occurs on a signaling link, the message (or set of messages) is retransmitted. MTP Level 2 is equivalent to the OSI Data Link Layer.

MTP Level 3 provides message routing between signaling points in the SS7 network. MTP Level 3 re-routes traffic away from failed links and signaling points and controls traffic when congestion occurs. MTP Level 3 is equivalent to the OSI Network Layer.

ISDN User Part (ISUP)

The ***ISDN User Part*** (***ISUP***) defines the protocol used to set-up, manage, and release trunk circuits that carry voice and data between terminating line exchanges (e.g., between a calling party and a called party). ISUP is used for both ISDN and non-ISDN calls. However, calls that originate and terminate at the same switch do not use ISUP signaling.

Telephone User Part (TUP)

The ***Telephone User Part*** (***TUP***) is used to support basic call setup and tear-down.

Signaling Connection Control Part (SCCP)

SCCP provides connectionless and connection-oriented network services and global title translation (GTT) capabilities above MTP Level 3. A global title is an address (e.g., a dialed 800 number, calling card number, or mobile subscriber identification number). SCCP is used as the transport layer for TCAP-based services.

Transaction Capabilities Applications Part (TCAP)

TCAP supports the exchange of non-circuit related data between applications across the SS7 network using the SCCP connectionless service. Queries and responses sent between SSPs and SCPs are carried in TCAP messages. In mobile networks (IS-41 and GSM), TCAP carries Mobile Application Part (MAP) messages sent between mobile switches and databases to support user authentication, equipment identification, and roaming.

Operations, Maintenance and Administration Part (OMAP) and ASE

OMAP and ***ASE*** are areas for future definition. Presently, OMAP services may be used to verify network routing databases and to diagnose link problems.

Technical words and phrases

PBX　专用小交换机，专用交换分机

telephony　[ti'lefəni]　*n.*　电话机制造法，通话法

external　[eks'tə:nl]　*adj.*　外部的，客观的，表面的

dedicated　['dedikeitid]　*adj.*　专用的

premise ['premis] *n.* 前提，房屋

trunk line [trʌŋk] [lain] *n.* 干线，中继线

conversely ['kɔnvəːsli] *ad.* 相反地

switch [switʃ] *v.* 交换

configuration [kənfigju'reiʃən] *n.* 配置

transmission [trænz'miʃən] *n.* 传送，传播

protocol ['prəutəkəl] *n.* 协议

Ethernet 以太网

packet-switched 分组交换，包交换

hybrid ['haibrid] *n.* 混合物

circuit-switched 电路交换

reside [ri'zaid] *v.* 居住，留驻，存在，属于

legacy ['legəsi] 传统

utility [juː'tiliti] *n.* 公用程序，公用事业

off hook 摘机

call forwarding 呼叫前转

flexibility [fleksi'biliti] *n.* 柔韧性，机动性，灵活性

administer [əd'ministə] *v.* 给予，施予；用；施行，进行；管理；提出；有助于

broadband ['brɔːdbænd] 宽带

facility [fə'siliti] *n.* 设施，设备

packetize 分成包； 包格式化

signaling ['signəliŋ] *n.* 信令

component [kəm'pəunənt] *n.* 组件，组成部分；元件

body ['bɔdi] *n.* 实体

appropriate [ə'prəuprieit] *adj.* 适当的，合适的

specific [spi'sifik] *adj.* 特殊的，明确的，具有特效的，特定的，具体的

inefficient [ini'fiʃənt] *adj.* 无效率的，无能的

International Telecommunications Union （ITU）国际电信联盟

in-band 带内

out-of-band 通带外，带外

traffic ['træfik] *n.* 流量，通信量

installation [instə'leiʃ(ə)n] *n.* 安装，安置；装置，设备；设施

signaling transfer points（STP） 信令转接点

signaling points（SP）信令点

LAN（LOCAL AREA NETWORK） 局域网

bi-directional *adj.* 双向作用的，双向的

capability [keipə'biliti] *n.* 能力，性能，容量

exchange [iks'tʃeindʒ] *n.* 交换，调换，交换机

Intelligent Network（IN） 智能网
fraudulent ['frɔ:djulənt] *adj.* 欺诈的，不正的，不诚实的
SSP (Service Switching Point) 信令交换点
SCP (Service Control Point) 信令控制点
flow control 流量控制
setup ['setʌp] *n.* 建立
identify [ai'dentifai] *vt.* 识别，认明，鉴定
characteristics *n.* 特性
validation [væli'deiʃən] *n.* 确认
congestion [kən'dʒestʃən] *n.* 堵塞，充满，拥挤；阻塞（常发生于超载时）
equivalent [i'kwivələnt] *adj.* 相等的，相当的，同意义的
tear-down 拆除，拆毁
global title 全局标题

7.2 Reading Materials

1. World Telecommunication Standardization Assembly outlines future global standards-setting

The World Telecommunication Standardization Assembly drew to a close today — World Standards Day[1] — with a plan for future global standards-setting and a clear statement about the direction of the future work of ITU's Telecommunication Standardization Sector (ITU-T)[2]. Internet-related issues and next-generation networks emerged as key areas. 475 delegates from 75 countries participated.

The 8-day meeting covered a wide range of issues that will impact the future direction of the telecommunication industry. It made important decisions that lay the foundations for the next generation of information and communication technologies. The Assembly also streamlined the ITU-T work programme to achieve greater efficiency in the production of ITU standards (ITU-T Recommendations).

"This WTSA has seen much intense deliberation," said Mr Roberto Blois, Deputy Secretary-General. "We have always to expect that there will be some difference of opinion. The fact that we have resolved these issues is testament to the value and power of ITU as an able architect of the standards that underpin the world's communications networks," Blois said.

"This Assembly made a great step forward in establishing a study group to deal with next-generation networks," said Mr José Leite Pereira Filho, Member of the Board of Directors of Anatel on behalf of the host country. Leite added that another key achievement of this Assembly has been the approval of a new resolution, along with a detailed action plan, aimed at bridging the standardization gap between developed and developing countries. He also stressed the importance of the consensual decision-making process. "The path you went through was very laborious, requiring unselfishness, understanding and an overwhelming disposition of cooperation", he said.

Mr Houlin Zhao, Director of ITU's Telecommunication Standardization Bureau (TSB) commended the Chairman for his leadership, his ability to steer the work of the Assembly to a successful conclusion and for having achieved sound results consensually, "We agreed new tools, resolutions, decisions and guidelines that will make ITU-T more efficient and much stronger." Zhao told delegates.

2. IP PBXs: Emerging Into Dominance

Enterprise VoIP is thriving. Worldwide IP PBX line shipments (including IP-enabled and server-based) will exceed traditional PBX line shipments for the first time in 2005. While the total PBX market is forecast to grow by a compound annual growth rate of 6.6% through 2009, the traditional PBX is in rapid decline while the IP PBX continues to gain momentum throughout the forecast period. This report identifies market trends that are driving the growth of the IP PBX. It also shows vendor market shares for 2004 and 1Q 2005.

3. What are the advantages of an IP-PBX system?

Multiple Branch Offices

With VoIP, expanding your business phone system to multiple branch office sites is easy. Integrated IP Gateways allow you to traffic calls between offices over the Internet and save on long distance charges. Make certain that your IP-PBX has an administration tool that simplifies the process of configuring IP gateways between remote systems.

Toll Bypass

IP-PBXs enable businesses to reduce the cost of long distance calling by routing calls inexpensively over IP networks. If you have overseas facilities, using an IP-PBX could reduce your business's costs significantly.

4. What is Signaling?

Signaling refers to the exchange of information between call components required to provide and maintain service.

As users of the PSTN, we exchange signaling with network elements all the time. Examples of signaling between a telephone user and the telephone network include: dialing digits, providing dial tone, accessing a voice mailbox, sending a call-waiting tone etc.

SS7 is a means by which elements of the telephone network exchange information. Information is conveyed in the form of messages.

5. What is Intelligent Network?

Intelligent Network (IN) is a telephone network architecture originated by Bell Communications Research (Bellcore) in which the service logic for a call is located separately from the switching facilities, allowing services to be added or changed without having to redesign switching equipment. According to Bell Atlantic, IN is a"service-specific" architecture. That is, a certain portion of a dialed phone number, such as 800 or 900, triggers a request for a specific service. A later version of IN called Advanced Intelligent Network (AIN) introduces the idea of a "service-independent" architecture in which a given part of a telephone number can interpreted differently by different services depending on factors such as time of day, caller identity, and type of call. AIN makes it easy to add new services without having to install new phone equipment.

Bellcore called its network IN/1. It included this model:

- The customer's telephone
- The switching system
- A database called a service control point (SCP) that defines the possible services and their logic
- A service management system (SMS)

6．What is Advanced Intelligent Network?

The Advanced Intelligent Network (AIN) is a telephone network architecture that separates service logic from switching equipment, allowing new services to be added without having to redesign switches to support new services. It encourages competition among service providers since it makes it easier for a provider to add services and it offers customers more service choices.

Developed by Bell Communications Research, AIN is recognized as an industry standard in North America. Its initial version, AIN Release 1, is considered a model toward which services will evolve. Elsewhere, the International Telecommunications Union (see ITU-T), endorsing the concepts of AIN, developed an equivalent version of AIN called Capability Set 1 (CS-1). It comes in evolutionary subsets called the Core INAP capabilities.

7．How Advanced Intelligent Network Works?

Briefly, here's how AIN Release 1 works:

- A telephone caller dials a number that is received by a switch at the telephone company central office.
- The switch - known as the Service Switching Point (SSP) - forwards the call over a Signaling System 7 (SS7) network to a Service Control Point (SCP) where the service logic is located.
- The Service Control Point identifies the service requested from part of the number that was dialed and returns information about how to handle the call to the Service Switching Point.
- In some cases, the call can be handled more quickly by an Intelligent Peripheral (IP) that is attached to the Service Switching Point over a high-speed connection. For example, a customized voice announcement can be delivered in response to the dialed number or a voice call can be analyzed and recognized.
- In addition, an "adjunct" facility can be added directly to the Service Switching Point for high-speed connection to additional, undefined services.

One of the services that AIN makes possible is Local Number Portability (LNP).

7.3 Exercises

1．Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）传真机
（2）调制解调器
（3）综合业务数字网

（4）互联网浏览器
（5）中继线
（6）呼叫转移
（7）5 类交换机
（8）业务逻辑
（9）呼叫
（10）信令
（11）带内
（12）带外
（13）信令交换点
（14）智能网

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）A PBX is a telephony device that is used by most （　　）- and （　　）-sized companies.
（2）A PBX is a much less expensive solution than giving each user in a business a dedicated external telephone line. A PBX can connect （　　）,（　　）（　　）, and many other communication devices.
（3）Internal calls are routed or switched by the（　　）between telephones that are physically connected to the PBX.
（4）An IP PBX is a Private Branch eXchange（　　）that supports the（　　）protocol to connect phones by using an Ethernet or（　　）-switched LAN and sends its voice conversations in（　　）packets.
（5）（　　）data and （　　）networks are One of the main advantages of an IP PBX.
（6）The two essential components to all telephone calls are actual（　　）and the information that instructs telephone exchanges to establish connections and（　　）the “content” to an appropriate（　　）.
（7）SS7 is simply a set of protocols that describe a means of（　　）between telephone switches in public telephone networks.
（8）PSTN is the abbreviation of（　　）.
（9）The signaling network which carries the（　　）control traffic. It is a（　　）-switched network using a common channel switching protocol.
（10）The primary function of SS#7 is to provide（　　）control, remote（　　）management, and（　　）capabilities for the inter- office telephone network.
（11）STP is the abbreviation of（　　）.
（12）Out-of-band signaling provides faster（　　）setup times, support for（　　）services and improved（　　）over fraudulent network usage.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（　　）The PBX equipment is typically installed at a business's premises and connects calls between the telephones located and installed in the business site.

（2）（　　）Trunk lines are typically available for making and receiving calls that are external to the business from an external source such as the PSTN.

（3）（　　）The calls placed between users within the business require special dialing digits or use of an external trunk line.

（4）（　　）PBXs contain line cards that support various transmission protocols such as ISDN, T1/E1, HDSL, and ADSL.

（5）（　　）A hybrid IP PBX supports the IP protocol for sending voice conversations in packets, but also connects traditional analog and digital circuit-switched TDM telephones.

（6）（　　）An IP PBX is telephone switching equipment that resides in a telephone company.

（7）（　　）IP PBX can also reduce long-term operation and maintenance costs.

（8）（　　）Administrators can easily configure their services by using an Internet browser or another IP-based utility on IP PBX.

（9）（　　）Centrex frees the customer from the costs and responsibilities of major equipment ownership.

（10）（　　）In IP telephony, voice conversations are analogue and transmitted across the network.

（11）（　　）The principal organization with responsibility for SS7 administration is the ETSI.

（12）（　　）In the past, in-band signaling techniques were used on interoffice trunks.

（13）（　　）The circuit-switched "user" network which actually carries the user voice and data traffic.

（14）（　　）The circuit-switched "user" network provides a physical path between the source and destination.

（15）（　　）SS#6 is the defined interoffice signaling protocol for ISDN.

（16）（　　）Signaling transfer points handle the SS#7 control network as well as the user circuit-switched network.

（17）（　　）The STPs route SS#7 control packets across the signaling network.

（18）（　　）Signaling occurs in-band on voice channels.

（19）（　　）SCCP provides connectionless and connection-oriented network services and global title translation (GTT) capabilities above MTP Level 2.

7.4　课文参考译文　程控交换系统及其信令

7.4.1　用户交换机

用户交换机（PBX）是主要为中等规模或大公司使用的一种电话设备，可以让用户共享几条外线拨打电话。PBX 作为一种廉价的解决方案，给予商务用户一条专用外线。电话机、传真机、调制解调器和许多其他的通信设备，都可以连接到 PBX。

PBX 通常安装在企业内部，为企业内部的呼叫建立连接。一定数量的外线，也称为主干线，通常用来处理企业与外网（如公众交换电话网 PSTN）之间的呼叫。

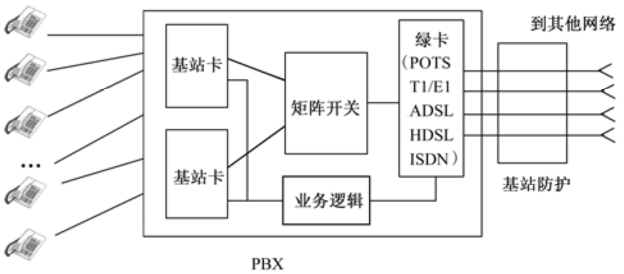
由企业内部向外网发起的呼叫通常在外线号码之前加拨一个 0 或 9，连接外网的主干线将自动完成本次呼叫。相反，由于企业内部呼叫的路由或交换都是由 PBX 完成的，因而不需要加拨特殊号码或使用主干线。

在中等规模和大型的公司，PBX 的配置方式通常如下：

- 单独的一个 PBX 将支持所有类型的商务呼叫
- 配置一组 PBX，这几个 PBX 可能互相分离，也有可能实现互联。

7.4.2　PBX 系统

PBX设备安装一种称为“线卡”的装置，该装置支持各种传输协议，如ISDN、T1/E1、HDSL和ADSL（如图7-1所示）。PBX同样支持POTS（传统电话业务）功能，该功能使得PBX设备在发生停电的情况下仍旧可以拨打外线电话。



译图 7-1　PBX 系统框架

7.4.3　IP PBX

IP PBX 就是支持 IP 协议的 PBX 设备，IP PBX 通过以太网或基于分组交换的局域网发起呼叫，以 IP 数据包的形式传输语音信号。多功能制式的 IP PBX 除了可以将语音信号转换成 IP 数据包后进行传输，同样也支持传统的模拟电话和基于电路交换的数字式电话（TDM 制式）。IP PBX 设备安装在企业内部而不是电信运营商那里。

IP PBX 的主要优势之一是它将数据和语音融合在一起。这就意味着互联网接入（包括 VoIP 通信和传统的电话业务）将可以使用一条主干线路连接用户。随着企业的发展，这种融合带来了灵活性，长期而言也可以降低运营和维护的成本。与传统的 PBX 设备一样，IP PBX 的所有权属于企业。

比起传统的 PBX， IP PBX 更易于管理。因为系统管理员可以使用互联网浏览器，或基于 IP 的软件工具来配置 IP PBX 业务，且无需另外安装线缆和配线架。使用 IP PBX 后，搬移一部

IP 电话机非常简单，只需要把话机拔下，然后插到新的地方即可。之外，安装 IP PBX 的企业也不需要增加基础设施方面的开销，也就是说，电路交换设备和分组交换设备的管理维护是一起进行的，而不是分开的。因此，IP PBX 业务比传统的 PBX 更易于管理。

7.4.4 Centrex

虚拟交换机（Centrex）是一种特殊的商用解决方案（主要用于语音业务）。呼叫控制和业务逻辑功能均由业务运营商维护，因此，通常安装在业务供应商一侧。Centrex 使得用户不再需要购买和维护设备，因此 Centrex 被认为是一种外包解决方案。

呼叫控制和业务逻辑主要是指处理一个电话呼叫并提供该呼叫的业务特征所需的功能。下面是呼叫控制和业务逻辑的几个例子：

- 识别对方已摘机，提示播放拨号音
- 剖析呼叫号码，以决定该呼叫的路由终点
- 决定被叫方是否可用或正忙，是否进行呼叫前传、以便采取相应的措施。（比如播放振铃音、呼叫忙音或呼叫等待音，将呼叫转发到语音邮箱，或将呼叫前转到第三方）
- 识别被叫方何时应答，主叫方或被叫方何时挂机，并记录相应的计费信息

在传统的 Centrex 业务里（比如模拟 Centrex 和 ISDN Centrex），呼叫控制和业务逻辑放置在交换局的 5 类交换机内部。5 类交换机也负责电信号的传输和交换，这些电信号携带着呼叫方的语音或其他信息（比如传真）。

7.4.5 IP Centrex

在 IP 电话里，语音信号被数字化后转换成数据包在网络上传输。IP Centrex 主要指一系列 IP 电话的解决方案，用户使用 Centrex 业务向网络发起呼叫，语音被打成数据包在宽带接入设备上传输。IP Centrex 继承了传统 Centrex 业务的优点，同时也带来了 IP 电话的优势。IP 电话的优势之一就是带宽利用率的提高。在 IP Centex 里，一台宽带接入设备将为同时发起的大量呼叫传输分组化语音数据流。当没有呼叫发起时，将有更多的可用带宽在局域网上传输高速数据。

7.4.6 信令系统

电话呼叫有两个重要的要素。第一点，也是最显著的，是通话的内容——如语音、传真和调制解调器数据等。第二点是用于完成交换过程的信息，即建立连接，将通话内容路由到相应的目的地的过程。电话信令就是为前面提到的第二个要素制定相应的规范，以实现第一点要素的内容，这些规范被称为协议。7 号信令或 7 号信令系统就是一组协议，该协议用于描述在公众电话网络里电话的交换过程。7 号信令由世界各地的不同实体分别制定，故其内容在不同的地方有一些特殊的变化，但都由国际电信联盟（ITU-T）（统一负责）组织进行 7 号信令的管理。

7 号信令系统（SS#7 or C7）是由电话运营商用于局间信令的。过去，带内信令的技术主要用于局间中继线上，这种信令方式在呼叫控制和实际连接的呼叫中使用相同的物理路径。该信令方式效率不高，正逐渐地被带外信令或公共信道信令技术所代替。

要理解 7 号信令，首先必须理解公众交换电话网（PSTN）原来所用信令机制的不足之处，即效率低下。迄今为止，绝大部分电话连接任务还是由以带内信令为主的一系列技术来实现。

使用公共信道信令的网络是由两个网络合并起来的：

第一，用户电路交换网承载用户的语音和数据流量。这在源和目的地之间提供了一条物理路径。

第二，信令网承载呼叫控制的信息。这是一个基于分组交换的网络，使用公共信道交换协议。

以前的公共信道局间信令协议是基于 6 号信令系统（SS#6）的，目前（已被）7 号信令（所取代），普遍应用于全球各地的通信设备中。7 号信令定义了 ISDN 的局间信令协议，并广泛用于 ISDN 以外的环境中。

7 号信令的主要功能是为局间电话网络提供呼叫控制、远程网络管理和维护功能。7 号信令通过在 7 号信令话务交换机(信令点或 SP)和 7 号信令转接点（STP）之间交换控制信息实现上述功能。

交换局（SP）（的概念）用于针对 7 号信令网和用户电路交换网。原则上，7 号信令将其在电路交换网上建立的连接路径告知交换局，使信令转接点可以在信令网中对 7 号信令控制的数据分组进行路由。交换局可能是信令转接点，但也有可能不是。

信令链路

7 号信令在 56 或 64Kbps 的双向信道上实现交换功能，该双向信道被称为信令链路，信令产生在带外专用信道上而不是带内语音信道上。与带内信令相比，带外信令提供以下一些功能：

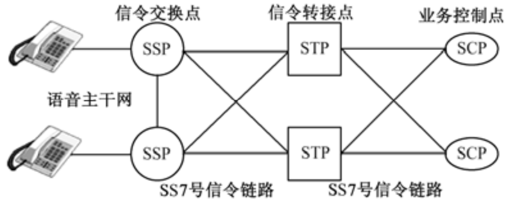
- 更高速的呼叫建立时间
- 更高效语音的电路利用率
- 支持智能网（IN）业务，智能网业务需要在网络元素之间实现信令路由，而不需要语音中继电路（比如数据库系统）
- 改进的控制能力用于防范网络欺诈

信令点

7 号信令网中的每一个信令点由一个数字点码唯一标识。该数字点码携带于信令中，在信令点之间的交换中识别每一条信令的源和目的地。每一个信令点使用一张路由表来为信令选择合适的路径。

7 号信令网中有三种信令点（如图 7-2 所示）

- SSP（业务交换点）
- STP（信令转接点）
- SCP（业务控制点）



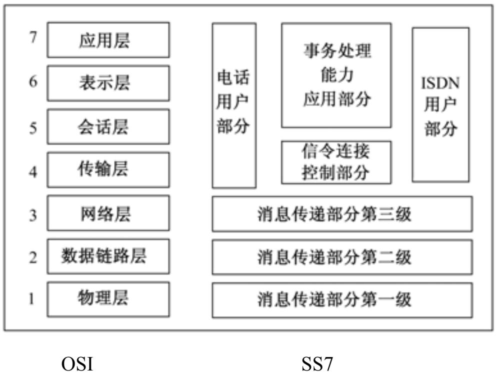
译图 7-2　7 号信令网

7 号信令协议栈

7 号信令协议的硬件和软件功能被划分成称为“级”的功能实体。这些“级”对应于开放式系统互联（OSI）中的 7 层模型，该模型由国际标准化组织（ISO）定义，具体如图 7-3 所示。

消息传递部分（MTP）

消息传递部分由三级功能组成。最低一级，即 MTP 第一级相当于 OSI 的物理层，它定义了数字信令链路的物理、电气和功能特性。物理接口定义 E-1（2048 Kbps; 信道速率 32 64 Kbps）、DS-1 (1544 Kbps; 信道速率 24 64 Kbps)、V.35 (64 Kbps)、DS-0 (64 Kbps)和 DS-0A (56 Kbps)。



译图 7-3　OSI 参考模型与 7 号信令协议栈

MTP 第二级保证信令链路中信令消息的正确传输。第二级实施流量控制、消息序列验证和差错检测。当信令链路中发生错误时，消息（或一组消息）将被重发。MTP 第二级相当于 OSI 的数据链路层。

MTP 第三级保证 7 号信令网中信令点之间信令消息的路由。第三级绕开有故障的链路、信令点重新路由，当堵塞发生时进行流量控制。MTP 第三级相当于 OSI 的网络层。

ISDN 用户部分（ISUP）

ISDN 用户部分（ISUP）定义了在线路的两个终端之间（主叫方和被叫方）用来建立呼叫、管理和释放承载语音和数据的中继电路的协议。ISUP 同时用在 ISDN 和非 ISDN 呼叫上。然而，在同一个交换机发起和终止的呼叫不使用 ISUP 信令。

电话用户部分（TUP）

电话用户部分（TUP）用于支持基本呼叫建立和拆除。

信令连接控制部分（SCCP）

SCCP 提供在 MTP 第三级上的无连接和面向连接的网络业务和全局码（GTT）转换功能。全局码是一个地址（例如 800 号码、电话卡号码或移动用户标识码）。SCCP 用于给基于事务处理能力应用部分（TCAP）的业务提供传输层。

事务处理能力应用部分（TCAP）

TCAP 支持 7 号信令网中应用程序之间非电路相关信息的交换，使用 SCCP 无连接业务。SSP 和 SCP 之间查询和响应通过 TCAP 信令消息来发送。在移动网络(IS-41 和 GSM)中，TCAP 在移动交换机和数据库之间传输移动应用部分（MAP）消息，支持用户的鉴权、设备识别和漫游。

运行、维护和管理部分（OMAP）和 ASE

OMAP 和 ASE 用于定义未来事务。目前的 OMAP 业务用于验证网络路由数据库和诊断链路故障。

Unit 8 3G Overview

8.1 Text

8.1.1 Evolution of Mobile Radio Communication

A brief history of the evolution of mobile communications throughout the world is useful in order to appreciate the enormous impact that cellular radio and Personal Communication Services (PCS) will have on all of us over the next several decades. It is also useful for a newcomer to the cellular radio field to understand the tremendous impact that government regulatory agencies and service competitors wield in the evolution of new wireless systems, services, and technologies.

Wireless communications is enjoying its fastest growth period in history, due to enabling technologies which permit widespread deployment. The recent exponential growth in cellular radio and personal communication systems throughout the world is directly attributable to new technologies of the 1970s, which are mature today. The future growth of consumer-based mobile and portable communication systems will be tied more closely to radio spectrum allocations and regulatory decisions which affect or support new or extended services, as well as to consumer needs and technology advances in the signal processing.

Many mobile radio standards have been developed for wireless systems throughout the world, and more standards are likely to emerge. The evolution of mobile radio communication systems is shown in Figure 8-1.

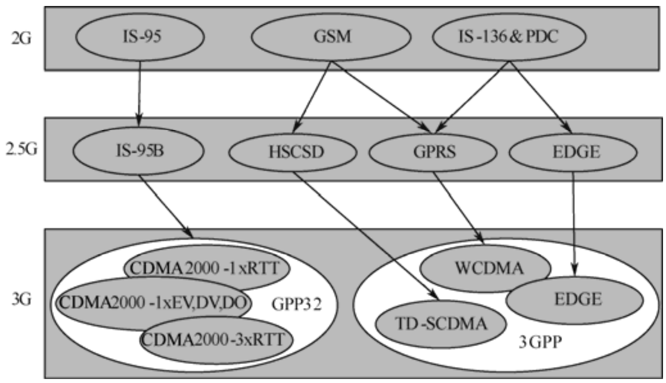


Figure 8-1 Evolution of mobile radio communication systems

8.1.2 3G Standards

3G is a short term for third-generation wireless, and refers to near-future developments in personal and business wireless technology, especially mobile communications. This phase is expected to reach maturity between the years 2003 and 2005.

The third generation, as its name suggests, follows the first generation (1G) and second generation (2G) in wireless communications. The 1G period began in the late 1970s and lasted

through the 1980s. These systems featured the first true mobile phone systems, known at first as "cellular mobile radio telephone. " These networks used analogue voice signalling, and were little more sophisticated than repeater networks used by amateur radio operators. The 2G phase began in the 1990s, and much of this technology is still in use. The 2G cell phone features digital voice encoding. Examples include CDMA, TDMA, and GSM. Since its inception, 2G technology has steadily improved, with increased bandwidth, packet routing, and the introduction of multimedia. The present state of mobile wireless communications is often called 2.5G.

Ultimately, 3G is expected to include capabilities and features such as:

- Enhanced multimedia (voice, data, video, and remote control)
- Usability on all popular modes(cellular telephone,e-mail,paging,fax,videoconferencing, and Web browsing)
- Broad bandwidth and high speed (upwards of 2 Mbps)
- Routing flexibility (repeater, satellite, LAN)
- Operation at approximately 2 GHz transmit and receive frequencies
- Roaming capability throughout Europe, Japan, and North America

While 3G is generally considered applicable mainly to mobile wireless, it is also relevant to fixed wireless and portable wireless. The ultimate 3G system might be operational from any location on, or over, the earth's surface, including use in or by homes、businesses、government offices、medical establishments、military、personal and commercial land vehicles、portable、space stations and spacecraft, et als.

Proponents of 3G technology promise that it will "keep people connected at all times and in all places." Researchers, engineers, and arketers are faced with the challenge of accurately predicting how much technology consumers will actually be willing to pay for. (Recent trends suggest that people sometimes prefer to be disconnected, especially when on vacation.) Another concern involves privacy and security issues. As technology becomes more sophisticated and bandwidth increases, systems become increasingly vulnerable to attack by malicious hackers (known as crackers) unless countermeasures are implemented to protect against such activity.

CDMA2000 and UMTS were developed separately and are 2 separate ITU approved 3G standards. CDMA2000 1xRTT, CDMA2000 1xEV-DO (Evolution, Data Only) and future CDMA 2000 3x were developed to be backward compatible with cdmaOne. Both 1x types have the same bandwidth, chip rate and it can be used in any existing cdmaOne frequency band and network. Backward compatibility was a requirement for successful deployment for USA market. It is easy to implement because operators do not need new frequencies. UMTS was developed mainly for countries with GSM networks, because these countries have agreed to free new frequency ranges for UMTS networks. Because it is a new technology and in a new frequency band, whole new radio access network has to be build. The advantage is that new frequency range gives plenty of new capacity for operators.

1. CDMA2000

CDMA2000 specification was developed by the Third Generation Partnership Project 2 (3GPP2) , a partnership consisting of five telecommunications standards bodies: ARIB and TTC in

Japan, CWTS in China, TTA in Korea and TIA in North America. CDMA 2000 has already been implemented to several networks as an evolutionary step from cdmaOne as CDMA 2000 provides full backward compatibility with IS-95B. CDMA 2000 is not constrained to only the IMT-2000 band.

CDMA2000 represents a family of technologies that includes CDMA2000 1X and CDMA2000 1xEV.

The world's first 3G (CDMA2000 1x) commercial system was launched by SK Telecom (Korea) in October 2000. Since then, CDMA2000 1X has been deployed in Asia, North and South America and Europe, and the subscriber base is growing at 700,000 subscribers per day. CDMA2000 1xEV-DO was launched in 2002 by SK Telecom and KT Freetel. The commercial success of CDMA2000 has made the IMT-2000 vision a reality.

2. Universal Mobile Telecommunications System (UMTS)

Universal Mobile Telecommunications System (UMTS) is the European standard for 3G mobile communication systems which provide an enhanced range of multimedia services. It has evolved from its basic format through developments such as HSDPA (High Speed Downlink Packet Access) and HSUPA (High Speed Uplink Packet Access) to provide very high bandwidth capabilities to support the next generation of telecommunication services.

UMTS supports up to 1920 Kbps data transfer rates (and not 2 Mbit/s as frequently seen), although at the moment users in the real networks can expect performance up to 384 Kbit/s – in Japan upgrades to 3 Mbps are in preparation. However, this is still much greater than the 14.4 Kbps of a single GSM error-corrected circuit switched data channel or multiple 14.4 Kbps channels in HSCSD , and-in competition to other network technologies such as CDMA-2000, PHS or WLAN-offers access to the World Wide Web and other data services on mobile devices.

UMTS combines the W-CDMA air interface, GSM 's Mobile Application Part (MAP) core, and the GSM family of speech codecs.

Note that many wireless technologies use W-CDMA as their air interface, including FOMA and J-Phone .

Like other real-world W-CDMA implementations, UMTS uses a pair of 5 MHz channels, one in the 1900 MHz range for uplink and one in the 2100 MHz range for downlink. In contrast, the competing CDMA2000 system uses one or more arbitrary 1.25 MHz channels for each direction of communication. UMTS and other W-CDMA systems are widely criticized for their large spectrum usage, which has delayed deployment in countries that have not allocated new frequencies specifically for UMTS (such as the United States).

The specific frequency bands originally defined by the UMTS standard are 1885～2025 MHz for uplink and 2110～2200 MHz for downlink.

For existing GSM operators, it is a simple but costly migration path to UMTS: much of the infrastructure is shared with GSM, but the cost of obtaining new spectrum licenses and overlaying UMTS at existing towers can be prohibitively expensive.

3. TD-SCDMA

In China, GSM is the most popular wireless air interface standard, and the wireless subscriber

growth in China is unmatched anywhere in the world. For example, in late 2001 more than eight million cell phone subscribers were added in just one month in China alone! Given the huge potential market for wireless services in China, and China' desire to craft its own wireless vision. The China Academy of Telecommunications Technology (CATT) and Siemens Corporation jointly submitted an IMT-2000 3G standard proposal in 1998, based on Time Division-Synchronous Code Division Multiple Access (TD-SCDMA). This proposal was adopted by ITU as one of the 3G options in late 1999.

TD-SCDMA relies on the existing core GSM infrastructure and allows a 3G network to evolve through the addition of high data rate equipment at each GSM base station. TD-SCDMA combines TDMA and TDD techniques. Up to 384 Kbps of packet data is provided to data users in TD-SCDMA.

The radio channels in TD-SCDMA are 1.6 MHz in bandwidth and rely on smart antennas, spatial filtering, and joint detection techniques to yield several times more spectrum efficiency than GSM. A 5 millisecond frame is used in TD-SCMDA, and this frame is subdivided into seven time slots which are flexibly assigned to either a single high data rate user or several slower users. By using TDD, different time slots within a single frame on a single carrier frequency are used to provide both forward channel and reverse channel transmissions.

For the case of asynchronous traffic demand, such as when a user downloads a file, the forward link will require more bandwidth than the reverse link, and thus more time slots will be dedicated to providing forward link traffic than for providing reverse link traffic. TD-SCDMA proponents claim that the TDD feature allows this 3G standard to be very easily and inexpensively added to existing GSM systems.

8.1.3 Comparison of Wireless Communication Systems

Nowadays, the most popular wireless communication systems are TV Remote Control, Garbage Door Opener, Paging System, Cordless Phone and Cellular Phone systems in the following Table 8-1. For the purpose of comparison with these common household wireless remote devices, the major characteristics have been chosen and illustrated in Table 8-1, such as the types of service, level of infrastructure, cost, complexity required for the subscriber segment and base station segment etc..

Table 8-1 Comparison of Mobile Communication Systems

For Mobile Station							
Service		Coverage Range	Required Infrastructure	Complexity	Hardware Cost	Carrier Frequency	Functionality
TV Remote Control	MS	Low	Low	Low	Low	Infrared	Transmitter
	BS	Low	Low	Low	Low	Infrared	Receiver
Garbage Door Opener	MS	Low	Low	Low	Low	＜100 MHz	Transmitter
	BS	Low	Low	Low	Low	＜100 MHz	Receiver
Paging System	MS	High	High	Low	Low	＜1 GHz	Receiver
	BS	High	High	High	High	＜1 GHz	Transmitter

（续）

For Mobile Station							
Service		Coverage Range	Required Infrastructure	Complexity	Hardware Cost	Carrier Frequency	Functionality
Cordless Phone	MS	Low	Low	Moderate	Low	1～3 GHz	Transceiver
	BS	Low	Low	Low	Moderate	1～3 GHz	Transceiver
Cellular Phone	MS	High	High	High	Moderate	<2 GHz	Transceiver
	BS	High	High	High	High	<2 GHz	Transceiver

It is important to note that each of the five mobile radio systems given in Table 8-1 use a fixed base station. Virtually all mobile radio communication systems strive to connect a moving terminal to a fixed distribution system of some sort. For example, the receiver in the garage door opener converts the received signal into a simple binary signal which is sent to the switching center of the garage motor. Cordless telephones use fixed base stations so they may be plugged into the telephone line supplied by the phone company — the radio link between the cordless phone base station and the portable handset.

Notice that the expectations vary widely among the services, and the infrastructure costs are dependent upon the required coverage area. For the case of low power, hand-held cellular phones, a large number of base stations are required to insure that any phone is in close range to a base station within a city. If base stations were not within close range, a great deal of transmitter power would be required of the phone, thus limiting the battery life and rendering the service useless for hand-held users.

Because of the extensive telecommunications infrastructure of copper wires, microwave line-of-sight links, and fiber optic cables- all of which are fixed- it is highly likely that future hand-based mobile communication systems will continue to rely on fixed base stations which are connected to some type of fixed distribution system. However, emerging mobile satellite networks will require orbiting base stations. Data communication standards of current and emerging 2.5G and 3G are concretely listed in Table 8-2.

Table 8-2　2.5G /3G Data Communication Standards

Wireless Data Tech.	Channel BW	Duplex	Infrastructure change requires	New requirements	
				Spectrum	Handsets
HSCSD	200kHz	FDD	software upgrade at BS	No	Yes—New HSCSD handsets provide 56.7 Kbps on HSCSD networks and 9.6 Kbps on GSM networks with dual mode phones. GSM-only phones will not work in HSCSD networks.
GPRS	200kHz	FDD	new packet overlay including routers and gateways	No	Yes—New GPRS handsets work on GPRS networks at 171.2 Kbps, 9.6 Kbps on GSM networks with dual mode phones. GSM-only phones will not work in GPRS networks.
EDGE	200kHz	FDD	new transceiver at BS and software upgrades to BS controller and BS	No	Yes—New handsets work on EDGE networks at 384 Kbps, GPRS networks at 144 Kbps, and GSM networks at 9.6 Kbps with tri-mode phones. GSM and GPRS-only phones will not work in EDGE networks.

（续）

Wireless Data Tech.	Channel BW	Duplex	Infrastructure change requires	New requirements	
				Spectrum	Handsets
W-CDMA	5MHz	FDD	completely new BS	Yes	Yes—New W-CDMA handsets will work on W-CDMA at 2 Mbps, EDGE networks at 384 Kbps, GPRS network at 144 Kbps, GSM networks at 9.6 Kbps. Older handsets will not work in W-CDMA.
IS-95B	1.25MHz	FDD	new software in BS controller	No	Yes —New handsets will work on IS-95B at 64 Kbps and IS-95A at 14.4 Kbps. CdmaOne phones can work in IS-95B at 14.4 Kbps.
CDMA2000 1XRTT	1.25MHz	FDD	new software in backbone and new channel cards at BS, and a new packet service node	No	Yes—New handsets will work on 1xRTT at 144 Kbps, IS-95B at 64 Kbps, IS-95A at 14.4 Kbps. Older handsets can work in 1xRTT but at lower speeds.
CDMA2000 1xEV(DO and DV)	1.25MHz	FDD	software upgrade on 1Xrtt networks	No	Yes—New handsets will work on 1xEV at 2.4 Mbps, 1xRTT at 144 Kbps, IS-95B at 64 Kbps, IS-95A at 14.4 Kbps. Older handsets can work in 1xEV but at lower speeds.
CDMA2000 3xRTT	3.75MHz	FDD	backbone modifications and new channel cards at BS	Maybe	Yes—New handsets will work on 95A at 14.4 Kbps, 95B at 64 Kbps, 1xRTT at 144 Kbps, 3xRTT at 2 Mbps. Older handsets can work in 3X but at lower speeds.

Technical words and phrases

cellular ['seljulə] *adj.* 蜂窝的

signaling ['signəliŋ] *n.* 信号

space station 宇宙空间站，太空站

spacecraft ['speiskrɑ:ft] *n.* 宇宙飞船，宇宙航行器

pedestrian [pə'destriən] *n.* 行人，步行者

hiker ['haikə] *n.* 徒步旅行者

cyclist ['saiklist] *n.* 骑脚踏车的

camper ['kæmpə] *n.* 露营者

sophisticated [sə'fistikeitid] *adj.* 老练的，复杂的，尖端的

packet ['pækit] *n.* 包，分组，数据包

route [ru:t] *n.* 路由

paging [peidʃiŋ] *n.* 呼叫，寻呼

videoconference [vidiəu'kɔnfərəns] *n.* 视频会议

roam [rəum] *v.* 漫游

portable ['pɔ:təbl] *adj.* 可携带的，可搬运的，可移动的

malicious [mə'liʃəs] *adj.* 怀恶意的，恶毒的

countermeasure ['kauntəmeʒə] *n.* 对策

overlay [əuvə'lei] *v.* 覆盖，重叠

subscriber [səb'skraibə] *n.* 用户

disconnect [diskə'nekt] *v.* 断开

capacity [kə'pæsiti] *n.* 容量，能力

specification [spesifi'keiʃən] *n.* 规格，详述，详细说明书

delay [di'lei] *vt.* 延迟 *vi.* 耽搁，延误 *n.* 耽搁，延迟，耽搁

uplink ['ʌpliŋk] *n.* 上行链路

downlink ['daunliŋk] *n.* 下行链路

deployment [di'plɔimənt] *n.* 部署

operator ['ɔpəreitə] *n.* 操作员，行家，经营者，算子，运营商

migration [mai'greiʃən] *n.* 移民，移往，移动

prohibitively *adv.* 禁止， 起阻止作用，抑制

TV Remote Control TV 遥控

Garbage Door Opener 车库门开启器

Paging 寻呼

Cordless Phone 无绳电话

infrastructure ['infrəstrʌktʃə] *n.* 基础设施

distribution [distri'bju:ʃən] *n.* 分布

switching [switʃiŋ] *n.* 切换，交换

portable ['pɔ:təbl] *adj.* 可携带的，可搬运的，可移动的，便携的

battery ['bætəri] *n.* 电池，电池组

render ['rendə] *vt.* 致使

line-of-sight 视距

dual mode 双模

tri-mode 三模

HSCSD High Speed Circuit Switched Data 高速电路交换数据

GPRS（General Packet Radio Service） 通用分组无线电业务

EDGE Enhanced Data Rates for GSM Evolution 改进数据率 GSM 服务

8.2 Reading Materials

1．Nokia and Siemens to merge their communications service provider businesses

Nokia and Siemens today announced that they intend to merge the Networks Business Group of Nokia and the carrier-related operations of Siemens into a new company, to be called Nokia Siemens Networks. The 50-50 joint venture will create a global leader with strong positions in important growth segments of fixed and mobile network infrastructure and services.

The combined company is positioned to lead the development and implementation of revenue-generating and cost-saving products and services via its scale and global reach. Nokia Siemens Networks will have one of the world's best research and development teams with the ability to invest in next generation fixed and mobile product platforms and services. The new company will

have a world-class fixed-mobile convergence capability, a complementary global base of customers, a deep presence in both developed and emerging markets, and one of the industry's largest and most experienced service organizations.

"We believe the partnership with Siemens is the most effective way to build the scale and broad product portfolio necessary to compete globally and create value for shareholders," said Olli-Pekka Kallasvuo, CEO of Nokia. "The communications industry is converging, and a strong and independent Nokia Siemens Networks will be ideally positioned to help customers lower costs and grow revenue while managing the challenges of converging technology. " Olli-Pekka Kallasvuo will serve as chairman of Nokia Siemens Networks.

"This joint venture is an important step to strengthen our position in the market sustainably and to enable us to offer the best state of the art converged technologies and services to our customers," said Klaus Kleinfeld, CEO of Siemens. "This combination creates a leading industry player with immediate strength, excellent potential for growth and well-positioned to improve future profitability."

2．AT&T rolls out 3G cell network（By Chris Wetterich）

People who have AT&T phones and cellular cards for their laptop computers will be able to download information.

The company said Thursday the available speeds are "DSL-like," with downloads of 600 kilobits to 1,400 kilobits per second and uploads running at 500 to 800 kilobits per second.

To use the 3G network, customers must have LaptopConnect or a laptop embedded with 3G capability.

The total price for a two-year contract using a laptop or USB card is $59.99 a month, which does not include the cost of a laptop card, said AT&T spokesman Chris Comes. The cards can be purchased for as little as $49.99 on AT&T's Web site with a two-year service agreement.

The card and service are compatible with both the 3G network and AT&T's second-generation EDGE network. It is not the same technology as Wi-Fi, which does not involve the use of cellular towers and signals.

On AT&T phones, users can use 3G to see videos, games, pictures, music, entertainment, news and weather, according to the company.

AT&T provides the service in more than 200 major U.S. metropolitan areas.

Customers also can use "video share," a feature that lets those with phones equipped with a video camera send live video to another AT&T customer in an area with 3G service. That service costs $4.99 per month for 25 minutes, $9.99 for an hour or $0.35 per minute on a pay-per-use basis.

All future AT&T phones will be sold with the 3G technology, Comes said.

3．Singapore: mobile TV means rules for 3G

Singapore's Media Development Authority is proposing to bring content on existing third-generation mobile networks under the same rules as any future mobile TV operations. And both sectors may now have to meet existing broadcasting content requirements.

The regulator has outlined its proposals for how mobile TV will be overseen. One of its proposals is "to require both [mobile TV service] operators and cellular mobile TV operators to obtain broadcast services licenses before transmitting TV services over their networks."

To date, 3G operators have not been required to have their channels approved by the regulator. StarHub, SingTel and M1 all offer 3G TV channels. But the MDA said that it was taken a "platform-neutral" approach to all mobile TV services, so would need to "regulate mobile TV services equally regardless of the technology platform (e.g. 3G, DVB-H, DMB)".

"Current fixed TV services (eg. MediaCorp, SCV, SingTel) are regulated under the TV Programme Code, " said the MDA. "MDA proposes that mobile TV services be subjected to the same regime, ie. Television programmes offered on mobile TV should comply with the TV Programme Code."

4. Step into the 3G lifestyle!

I've never been much of a "techie," but keeping up with the times has been major priority of mine as technology after technology is churned out minute by minute.

The most accessible and handy gadget that people cannot live without is their mobile phone but it seems that there are a good lot of us that do not make best use of our handsets and their built in technology. Speaking for myself and a good handful of my friends, other than calling, texting and sending the occasional MMS, I never used a cellular phone for much else. That was until now!

LG Electronics, the worldwide technology leader in mobile communications, has introduced the LG-KU380, its newest feature-rich handset specially designed to provide affordable 3G technology to today's mobile phone-crazed generation. When we first heard of "3G technology," many of us believed it to be something expensive or difficult to use. LG has proved that assumption wrong.

The LG-KU380 is LG's second 3G handset to be chosen for GSM Association's "3G for All" campaign. This technology-packed handset is LG's answer to users' need for faster, feature-rich mobile services that are well within their budget and easy to use. You don't have to be a "techie" or "gadgeteer" to put this cellular phone to good use.

LG-KU380 offers enhanced mobile experience to users as it lets them enjoy faster downloads of up to 384 Kbps. Download music, video clips and other content plus enjoy other wireless multimedia services such video calling, Internet browsing, e-mail, real-time interactive gaming, and media streaming anytime, anywhere in less time.

The sleek, stylish handset lets users enjoy downloads as it supports external memory for storing video and music. You can also playback your favorite music in a variety of music files such as HF, MP3, AAC, AAC+, among others, or watch your favorite video clips in various video formats like MPEG4, 3GPP streaming, and 3GPP P D/L. For the shutterbug in you, LG-KU380 lets you click away with its 1.3 megapixel flash camera plus a VGA camera.

With all these features and a price tag which won't break your holiday budget, the LG-KU380 mobile phone is the ideal stocking stuffer for "gadget gurus" and "aspiring techies" alike this Christmas!

8.3 Exercises

1. Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）蜂窝
（2）路由
（3）漫游
（4）码片速率
（5）频带
（6）无线接入
（7）上行链路
（8）个人通信业务
（9）时隙
（10）寻呼系统
（11）移动台

2. Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）3G is a short term for （ ）wireless.
（2）The 1G networks used（ ）voice signaling.
（3）The 2G cell phone features（ ）voice encoding.
（4）CDMA2000 and UMTS were developed separately and are 2 separate（ ）approved 3G standards.
（5）3GPP2 is the abbreviation of （ ）.
（6）（ ） is the European standard for 3G mobile communication systems which supports up to（ ）Kbps data transfer rates
（7）UMTS combines the（ ）air interface, （ ）'s Mobile Application Part (MAP) core, and the（ ）family of speech codecs.
（8）PCS stands for the abbreviation of （ ）.
（9）The China Academy of Telecommunications Technology and Siemens Corporation jointly submitted an IMT-2000 3G standard proposal based on （ ）
（10）The radio channels in TD-SCDMA are（ ）MHz in bandwidth and rely on （ ）, （ ）, and（ ）techniques to yield several times more spectrum efficiency than GSM.

3. Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（ ）The 1G period began in the late 1970s and its networks used digital voice signalling,
（2）（ ）The 2G period began in the 1990s, and much of this technology is still in use.
（3）（ ）CDMA, TDMA, and GSM are all 3G technologies.

（4）（　　）Packet routing is employed in 2G technology.

（5）（　　）3G is generally considered applicable mainly to mobile wireless, however it is irrelevant to fixed wireless and portable wireless.

（6）（　　）CDMA2000 specification was developed by ITU.

（7）（　　）The world's first 3G commercial system was launched by SK Telecom in October 2000.

（8）（　　）The world's first 3G commercial system was based on CDMA2000 3x.

（9）（　　）The specific frequency bands originally defined by the UMTS standard are 1885-2025 MHz for downlink and 2110～2200 MHz for uplink.

（10）（　　）CDMA is the most popular wireless air interface standard in China.

（11）（　　）TD-SCDMA combines TDMA and FDD techniques.

（12）（　　）In TD-SCDMA system, up to 384 Kbps of packet data is provided to data users.

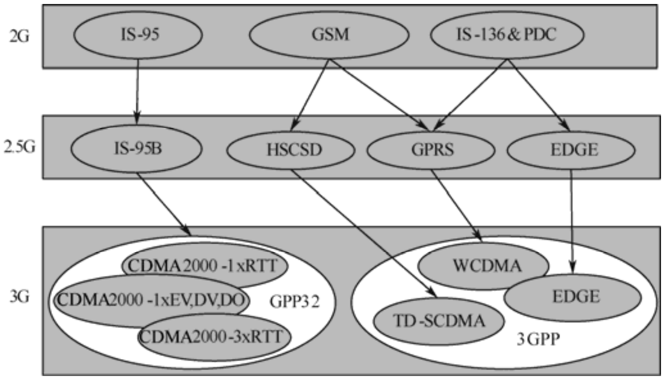
8.4　课文参考译文　3G 概览

8.4.1　移动无线通信系统的演进

移动通信的演化史对于我们去理解蜂窝无线通信和个人通信业务（PCS）在过去几十年里给予我们生活所带来的影响是非常有必要的。同时，无线通信简史对于一个刚刚踏入蜂窝无线领域的人来说，想要理解政府管理部门和业务竞争者对新的无线系统、业务和技术的影响力也是很有必要的。

无线通信系统的发展非常的快，这是因为相应技术使得无线通信系统的大范围部署成为了可能。近几年来全球蜂窝无线和个人通信系统领域指数级别的增长要归结于 20 世纪 70 年代诞生的新技术，这些技术在今天已完全成熟。未来，用户移动和手持通信系统的发展将与无线频谱的划分和无线管理规定两者之间的联系更为紧密。因为它们对新业务或拓展业务、用户需求和信号处理领域内的技术进展有着非常大的影响力。

全球无线系统中大多数标准已被开发出来，并且在将来很有可能出现新的无线标准。移动无线通信系统的演化见图 8-1 所示。



译图 8-1　移动无线通信系统的演进

8.4.2 3G标准

3G是第三代无线通信技术的简称，主要是指在不远的将来，在个人和商务方面应用的无线通信技术，尤其是移动通信方面。3G技术预期在2003—2005年之间达到成熟。

第三代无线通信技术，正如它的名字所隐含的，是紧随着第一代无线通信技术（1G）和第二代无线通信技术（2G）发展而来的。第一代无线通信技术始于20世纪70年代，并一直持续到20世纪80年代。第一代无线通信技术的特点是移动电话网络的首次出现，起初被称为“蜂窝移动无线电话”。该网络使用模拟语音信令，其复杂度与业余无线电操作员使用的中继网络差不太多。第二代无线通信技术的特点是数字化语音编码。像CDMA、TDMA和GSM都是属于第二代无线通信技术。自从2G诞生以来，其性能在不断地提高，包括改进的带宽、分组路由和多媒体技术的引入。当前使用的无线通信技术被称为2.5G。

3G最终将包含以下一些功能和特性：

- 增强型多媒体（语音、数据、视频和远程控制）
- 可以在所有大众化的模式下使用（包括蜂窝电话、电子邮件、寻呼、传真、视频会议和网页浏览）
- 高带宽和高速率（最高2Mbps）
- 灵活的路由方式（中继器、卫星、局域网）
- 运行在大概2GHz的速率下发送和接收信息
- 可在欧洲、日本和北美实现漫游功能

一般来说，3G主要应用在无线移动环境中，但它与固定无线和手持无线技术也有关系。将来，3G系统可以在地球上任何一个环境中得到应用，包括家庭、商务、政府机关、医疗机构、军事、个人和商用地面交通工具、手持设备、空间站和宇宙飞船等。

3G技术的倡导者承诺该技术将使“所有人在任何时候任何地点保持在线状态”。研究人员、工程师和市场营销人员现在所面临的问题是，消费者究竟会为哪些技术买单（最近的趋势表明人们更喜欢保持下线状态，尤其是在休假时期）。另一个问题涉及到了隐私和安全问题。随着技术变得更为复杂和带宽的不断增加，无线通信系统将会更加容易受到黑客的攻击，除非我们有相应的措施来保护系统免受这些攻击。

CDMA 2000和UMTS是两个分离开发的技术，这两个技术均得到了国际电信联盟的认可。CDMA 20001xRTT, CDMA 2000 1xEV-DO（EVolution, 只涉及数据）和未来的CDMA 2000 3x在开发中与cdmaOne保持了兼容性。所有的CDMA 2000 1x技术具有相同的带宽，码片速率，并可以在任何现有的cdmaOne频段和网络上运行。后向兼容性对于美国市场中来说是非常重要的。这样的话运营商无需增加新的频段，因此实施起来就容易得多。UMTS主要为部署GSM网络的国家而开发的，因为这些国家已达成协议为UMTS网络腾出新的频段出来。因为UMTS是一项新的技术并在新的频段上运行，所以需要建设全新的无线接入网络。但是这带来的优势也就是，新的频段为运营商带来了更多的用户容量。

1．CDMA2000

CDMA 2000 规范由第三代合作伙伴计划 2（3GPP2）开发，该组织由 5 个电信标准实体组成：ARIB 和 TTC（日本）、CWTS（中国）、TTA（韩国）和 TIA（北美）。作为cdmaOne 网络的进一步演化，CDMA2000 网络已经在部分地区得到部署。CDMA 2000 为 IS-95B 提供了全面的后向兼容性。CDMA2000 并不仅仅局限于 IMT-2000 的频段。

CDMA2000 代表了一系列的技术，包括 CDMA2000 1X 和 CDMA2000 1xEV。

世界上第一个 3G 商用系统（CDMA2000 1X）由 SK 电信于 2000 年 10 月在韩国部署。自那之后，亚洲、南北美和欧洲分别部署了 CDMA2000 1x 系统，用户数量以每天 70 万的速度增长。CDMA2000 1xEV-DO 由 SK 电信和 KT Freetel 于 2002 年投入运营。CDMA2000 在商业上的成功实施使得 IMT-2000 梦想的实现成为了现实。

2．通用移动通信系统（UMTS）

通用移动通信系统（**UMTS**）是由欧洲开发的3G移动通信技术，该技术支持增强型多媒体。UMTS由高速下行分组接入技术（HSDPA）和高速上行链路分组接入（HSUPA）发展而来，该技术提供了非常高的带宽以支持下一代电信业务。

UMTS 最高支持 1920 Kbps 数据传输速率（而不是我们通常看到的 2 Mbps），虽然目前用户在实际的网络里速率最高只能达到 384 Kbps——日本已经升级到 3 Mbps 为实现更高的速率做好准备。然而，这比 GSM 纠错电路交换数据网络中单一信道中所能达到的速率 14.4 Kbps、HSCSD（高速电路交换数据）中多重信道所能达到 14.4 Kbps 要快得多。这样，UMTS 技术才能与像 CDMA 2000 、PHS 或 WLAN 之类的技术相抗衡——这些技术提供在移动终端上接入万维网和其他数据业务。

UMTS 结合了 W-CDMA 空中接口，GSM 的移动应用部分（MAP）内核和 GSM 家族的语音编解码方案。

值得注意的是，许多无线技术使用 W-CDMA 作为其空中接口，包括 FOMA 和 J-Phone。

与其他实际的 W-CDMA 应用一样，UMTS 使用一对 5MHz 的信道，上行在 1900MHz 频段，下行在 2100MHz 频段。与之对应的，CDMA2000 为每对通信方提供一个或多个专用的 1.25 MHz 信道。UMTS 和其他的 W-CDMA 系统一直以来广受批判的是，它们均使用了大量的带宽，这阻碍了这些技术在一些国家的部署。在这些国家（如美国），并没有为 UMTS 分配了新的频段。

UMTS 标准原先定义的频段是上行 1885～2025 MHz、下行 2110～2200 MHz。

对于现有的 GSM 运营商来说，过渡到 UMTS 很简单但相当昂贵：UMTS 的许多设备是可以与 GSM 共享的，但需要获得新频段的牌照，并在现有基站上重叠 UMTS 网络，就这两方面来说，成本相当高。

3．TD-SCDMA

在我国，GSM 是最受欢迎的无线空中接口技术，无线用户在国内的增长速率在世界上也是最快的。以 2001 年为例，仅仅在中国就增加了 8000 多万手机用户。考虑到国内无线业务潜在的庞大市场，我国决定自行制定属于自己的无线标准。中国信息产业部电信科学技术研究院（CATT）和西门子公司在 1998 年联合提出了 IMT-2000 3G 标准，该标准基于时分同步的码分多址技术（TD-SCDMA）。该标准在 1999 年被国际电信联盟采纳为 3G 技术之一。

TD-SCDMA 充分利用现有的 GSM 设备，通过在每个 GSM 基站添加一台高速率设备来完成向 3G 网络的过渡。TD-SCDMA 结合了时分多址（TDMA）和时分双工（TDD）的技术。最高可以为用户提供 384Kbps 的数据速率。

TD-CDMA 采用 1.6MHz 带宽的无线信道，并依托智能天线、空域滤波和联合检测技术，使得其提供的带宽是 GSM 的好几倍。TD-SCDMA 的帧长为 5ms，每个帧包含 7 个时

隙。这些时隙可以被全部分配给一个高速率的用户或同时分配给几个低速率的用户。通过使用时分双工（TDD）技术，同一载波频率上每一帧内部的不同时隙可以同时提供前向信道或反向信道传输。

对于异步通信的需求，如在用户下载文件时，比起反向链路，前向链路将需求更多的带宽，因此与反向链路通信相比，也就意味着需要更多的时隙来提供前向链路通信。TD－SCDMA 的拥护者声称时分双工的特性使得该 3G 标准可以较容易并且廉价地融合进现有的 GSM 系统。

8.4.3 无线通信系统比较

如今最常见的无线通信系统是 TV 远程控制、自动车库门、寻呼系统、无绳电话和蜂窝无线系统，见表 8-1 所示。为了方便对下列常用的家用无线远程设备作一个比较，我们选择了几个要素，比如业务类型、基础设施要求、成本、用户设备以及基站的复杂度。

译表 8-1 不同移动通信系统比较

移动台							
业务		覆盖范围	基础设施要求	复杂度	硬件成本	载波频率	功能
TV 远程控制	MS	低	低	低	低	红外	发射机
	BS	低	低	低	低	红外	接收机
自动车库门	MS	低	低	低	低	<100MHz	发射机
	BS	低	低	低	低	<100MHz	接收机
寻呼系统	MS	高	高	低	低	<1GHz	接收机
	BS	高	高	高	高	<1GHz	发射机
无绳电话	MS	低	低	中	低	1～3GHz	收发机
	BS	低	低	低	中	1～3GHz	收发机
蜂窝电话	MS	高	高	高	中	<2GHz	收发机
	BS	高	高	高	高	<2GHz	收发机

值得注意的是，表 8-1 列出的 5 种移动无线系统均使用了固定基站。事实上，所有的移动无线系统都是将移动终端和某种分布式固定系统连接在一起的。自动车库门的接收机将接受信号转换为二进制信号，然后将该二进制信号发送至车库电机的交换中心。无绳电话搭配固定基座，因为这样，无绳电话就可以连接到电信运营商的通信电缆上，而无绳电话基座和手持终端之间的连接属于无线链路。

我们知道这几种业务是不尽相同的，同样，各种业务的基础设施建设成本取决于其覆盖范围。对于低功率蜂窝电话来说，则需要建设大量的基站来保证任何手机均在城市基站的覆盖范围之内。如果基站距离不是很近，那么就要求手机的发射功率很高，这样就限制了电池的寿命，使得蜂窝电话业务很难推向市场。

由于铜质电缆、微波视距链路和光缆等电信基础设施的存在（所有这些都是固定的），很有可能未来的手持移动通信系统将继续依赖固定基站，这些基站将连接到某种类型的分布式固定系统。不过，未来的移动卫星网络可能要求基站位于对地轨道上的。当前 2.5G 和未来 3G 移动通信系统的具体数据标准如表 8-2 所列。

译表 8-2　2.5G/3G 数据通信标准

<table>
<tr><th rowspan="2">无线数据技术</th><th rowspan="2">信道带宽</th><th rowspan="2">双工技术</th><th rowspan="2">基础设施需求</th><th colspan="2">新 的 需 求</th></tr>
<tr><th>频段</th><th>手 持 终 端</th></tr>
<tr><td>HSCSD</td><td>200kHz</td><td>FDD</td><td>基站软件升级</td><td>不需要</td><td>是——新的 HSCSD 手持终端在 HSCSD 网络上提供 56.7Kbps 的速率，在 GSM 网络上提供 9.6Kbps 的速率（需要双模手机）。只支持 GSM 模式的手机不能运行在 HSCSD 网络中</td></tr>
<tr><td>GPRS</td><td>200kHz</td><td>FDD</td><td>路由器和网关使用新分组重叠技术</td><td>不需要</td><td>是——新的 GPRS 手持终端在 GPRS 网络上提供 171.2Kbps 速率，在 GSM 网络上提供 9.6Kbps 的速率（需要双模手机）。只支持 GSM 模式的手机不能运行在 GPRS 网络中</td></tr>
<tr><td>EDGE</td><td>200kHz</td><td>FDD</td><td>基站内部新的收发器、基站控制器和基站软件的升级</td><td>不需要</td><td>是——新的手持终端在 EDGE 网络上提供 384Kbps 速率，在 GPRS 网络上提供 144Kbps 的速率, 在 GSM 网络上提供 9.6Kbps 的速率（需要三模手机）。只支持 GSM 模式和 GPRS 模式的手机不能运行在 EDGE 网络中</td></tr>
<tr><td>W-CDMA</td><td>5MHz</td><td>FDD</td><td>全新的基站</td><td>需要</td><td>是——新的 W-CDMA 手持终端在 W-CDMA 网络上提供 2 Mbps 速率，在 EDGE 网络上提供 384Kbps 的速率, 在 GPRS 网络上提供 144 Kbps 速率，在 GSM 网络上提供 9.6Kbps 的速率。老式手持终端不能运行在 W-CDMA 网络中</td></tr>
<tr><td>IS-95B</td><td>1.25MHz</td><td>FDD</td><td>基站控制器需更新的软件</td><td>不需要</td><td>是——新的手持终端在 IS-95B 上提供 64Kbps 的速率，在 IS-95A 上提供 14.4Kbps 的速率。CdmaOne 手机在 IS-95B 上可以提供 14.4 Kbps 的速率</td></tr>
<tr><td>CDMA2000 1xRTT</td><td>1.25MHz</td><td>FDD</td><td>骨干网需更新软件，基站需更新信道板和分组业务节点</td><td>不需要</td><td>是——新的手持终端在 1xRTT 上提供 144Kbps 的速率，在 IS-95B 上提供 64Kbps 的速率，在 IS-95A 上提供 14.4Kbps 的速率。老式的手机可以在 1xRTT 上运行，但速率较低</td></tr>
<tr><td>CDMA2000 1xEV(DO and DV)</td><td>1.25MHz</td><td>FDD</td><td>1xRTT 网络需软件升级</td><td>不需要</td><td>是——新的手持终端在 1xEV 上提供 2.4 Mbps 的速率，在 1xRTT 上提供 144Kbps 的速率，在 IS-95B 上提供 64Kbps 的速率，在 IS-95A 上提供 14.4Kbps 的速率。老式的手机可以在 1xEV 上运行，但速率较低</td></tr>
<tr><td>CDMA2000 3xRTT</td><td>3.75MHz</td><td>FDD</td><td>骨干网需要修改，基站需更新信道卡</td><td>待定</td><td>是——新的手持终端在 95A 上提供 14.4Kbps 的速率，在 95B 上提供 64Kbps 的速率，在 1xRTT 上提供 144Kbps 的速率，在 3xRTT 上提供 2 Mbps 的速率。老式的手机可以在 3X 上运行，但速率较低</td></tr>
</table>

Unit 9 Internet of Things

9.1 Text

9.1.1 Introduction of IoT

The Internet of Things (IoT) refers to uniquely identifiable objects (things) and their virtual representations in an Internet-like structure. The term Internet of Things was first used by Adam Baumgarten in 1999. The concept of the Internet of Things first became popular through the Auto-ID Center and related market analysts publications. Radio-frequency identification (RFID) is often seen as a prerequisite for the Internet of Things. If all objects and people in daily life were equipped with radio tags, they could be identified and inventoried by computers. However, unique identification of things may be achieved through other means such as barcodes or 2D-codes as well.

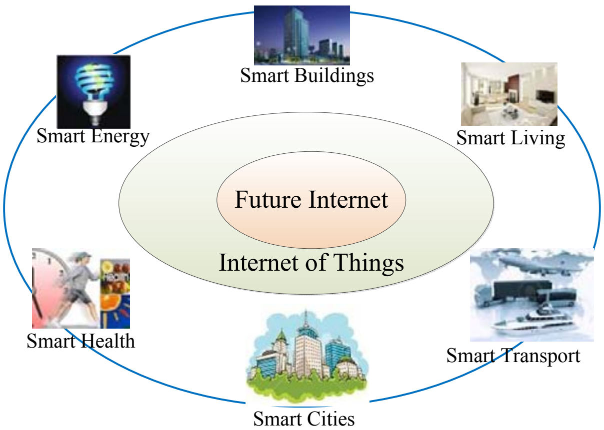


Figure 9-1 IoT and smart environments creation

Equipping all objects in the world with minuscule identifying devices could be transformative of daily life. For instance, business may no longer run out of stock or generate waste products, as involved parties would know which products are required and consumed. One’s ability to interact with objects could be altered remotely based on immediate or present needs, in accordance with existing end-user agreements.

Fields of applications include for example waste management, urban planning, sustainable urban environment, continuous care, emergency response, intelligent shopping, smart product management, smart meters, home automation and smart events. Alcatel-Lucent touchatag service and Violet’s Mirror gadget provide a pragmatic consumer oriented approach to the Internet of Things by which a developer can link real world items to the online world using RFID tags and QR Codes.

One key issue with the Internet of Things is the ability to rapidly create IoT applications. An approach taken by the Media and Graphics lab at the University of British Columbia (Canada)

focuses on a lightweight toolkit for developing IoT applications and targets rapid development using Web technologies and protocols. The toolkit has been described at the 2012 IoT (IEEE) conference and builds on previous IoT research, in particular the work on the MAGIC Broker as published at IoT 2010 (IEEE)

9.1.2 Smart City

Urban performance currently depends not only on the city's endowment of hard infrastructure ('physical capital'), but also, and increasingly so, on the availability and quality of knowledge communication and social infrastructure ('intellectual capital and social capital'). The latter form of capital is decisive for urban competitiveness. It is against this background that the concept of the smart city has been introduced as a strategic device to encompass modern urban production factors in a common framework and to highlight the growing importance of Information and Communication Technologies (ICTs), social and environmental capital in profiling the competitiveness of cities. The significance of these two assets - social and environmental capital - itself goes a long way to distinguish smart cities from their more technology-laden counterparts, drawing a clear line between them and what goes under the name of either digital or intelligent cities.

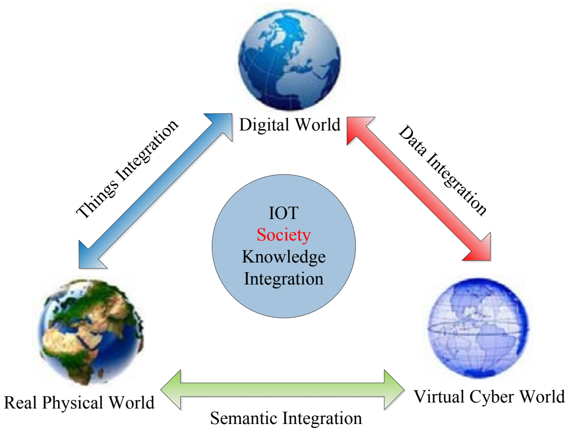


Figure 9-2 Internet of Things- a symbiotic interaction among the real/physical, the digital, virtual worlds and society

The concept of the smart city as the next stage in the process of urbanisation has been quite fashionable in the policy arena in recent years, with the aim of drawing a distinction from the terms digital city or intelligent city. Its main focus is still on the role of ICT infrastructure, but much research has also been carried out on the role of human capital/education, social and relational capital and environmental interest as important drivers of urban growth.

The European Union (EU), in particular, has devoted constant efforts to devising a strategy for achieving urban growth in a smart sense for its metropolitan city-regions. Other international institutions and think tanks also believe in a wired, ICT- driven form of development. The Intelligent Community Forum produces, for instance, research on the local effects of the worldwide ICT revolution. The OECD and EUROSTAT Oslo Manual stresses instead the role of

innovation in ICT sectors and provides a toolkit to identify consistent indicators, thus shaping a sound framework of analysis for researchers on urban innovation. At a mesoregional level, we observe renewed attention for the role of soft communication infrastructure in determining economic performance.

The availability and quality of the ICT infrastructure is not the only definition of a smart or intelligent city. Other definitions stress the role of human capital and education and learning in urban development. It has been shown, for example, that the most rapid urban growth rates have been achieved in cities where a high share of educated labor force is available.

Innovation is driven by entrepreneurs who innovate in industries and products which require an increasingly more skilled labor force. Because not all cities are equally successful in investing in human capital, an educated labor force – the 'creative class' – is spatially clustering over time. This tendency for cities to diverge in terms of human capital has attracted the attention of researchers and policy makers. It turns out that some cities, which were in the past better endowed with a skilled labor force, have managed to attract more skilled labor, whereas competing cities failed to do so. Policy makers, and in particular European ones, are most likely to attach a consistent weight to spatial homogeneity; in these circumstances the progressive clustering of urban human capital is then a major concern.

Wireless sensor networks is a specific technology that helps to create Smart Cities. The aim is to create a distributed network of intelligent sensor nodes which can measure many parameters for a more efficient management of the city. The data is delivered wirelessly and in real-time to the citizens or the appropriate authorities.

For example, citizens can monitor the pollution concentration in each street of the city or they can get automatic alarms when the radiation level rises a certain level. It is also possible for the authorities to optimize the irrigation of parks or the lighting of the city. Water leaks can be easily detected or noise maps can be obtained. Rubbish bins can send an alarm when they are close to being full.

Vehicle traffic can be monitored in order to modify the city lights in a dynamic way. Traffic can be reduced with systems that detect where the nearest available parking slot is. Motorists get timely information so they can locate a free parking slot quickly, saving time and fuel. This information can reduce traffic jams and pollution improve the quality of life.

9.1.3 Application of IoT—Control 4 Hospitality Solutions

From doors to drapes, Control4 hospitality solutions truly deliver the "wow factor" when it comes to the guest experience. From the moment hotel guests arrive, their stay can be choreographed to meet their personal preferences. Here are just a few examples of what is possible with Control 4:

A very special welcome

As guests approach their room or suite, the system recognizes if it is their first time in the room and "greets" them as they enter. Light fills the room, the curtains automatically open to reveal spectacular views and the TV displays a list of automated controls for guests to personalize.

One remote controls... everything

The TV can serve as a communications center, displaying messages regarding new voicemails, package deliveries and more. Guests can also control music, lights, thermostat, fireplace, drapes, do not disturb, and hotel services such as make up room, all from the TV GUI.

One-touch services

Guests can request services from the bellman's desk, valet, spa, golf club, front desk and more—right from a touch screen or the TV in their room. Hoteliers can choose from a wide range of services to offer your guests, while making back-end functions like mini-bar refills, housekeeping services and security settings more efficient to manage.

Customized scenes

Pre-set "scenes" make it easy for guests to match their mood or preference in one touch. For example, a "good night" button next to the bed can turn off the lights, TV and/or music; shut the curtains; and turn on the privacy notification for the room. Similarly, guests can personalize their wake-up routine—soft music or the morning news as the lights gently illuminate the room and the heat kicks on to the perfect temperature. When they're ready, drapes open to welcome the morning sun.

Make it easy for guests to "go green"

Enlist your guests' help by allowing them to press a "green" button to participate in the hotel's preferred green settings, automatically managing linen and towel service, temperature and lighting for maximum energy savings.

Technical words and phrases

Internet of Things (IoT)　物联网
Radio-frequency identification (RFID)　电子标签
inventory　['ɪnvəntri]　*n.*　存货清单；*vt.*　编制……的目录；总结
barcode　[bɑː'kəʊd]　条形码
2D-codes　二维码
minuscule　['minəskju:l]　*adj.*　非常小的，极不重要的
transformative　[træns'fɔ:mətiv]　*adj.*　有改革能力的，变化的，变形的
interact　[ˌɪntər'ækt]　*v.*　相互作用；互相影响；互动
sustainable　[sə'steɪnəbl]　*adj.*　可持续的；可以忍受的；可支撑的
intelligent　[in'telidʒənt]　*adj.*　有智力的；[计]智能的
pragmatic　[præg'mætɪk]　*adj.*　实际的；实用主义的；国事的
lightweight　['laɪtˌweɪt]　*adj.*　轻量的；不重要的 *n.* 轻量级选手；不重要的人
Information and Communication Technologies (ICTs)　信息通信技术
mesoregional　*adv.*　地方的，地域性的

metropolitan [ˌmetrə'pɔlitən] *adj.* 大都会的；大城市的；*n.*大城市人
The Intelligent Community Forum 智能社区论坛
framework ['freimwə:k] *n.* 构架；框架；（体系的）结构；机构，组织
renew [ri'nju:] *vt.* 重新开始；使更新；使恢复；补充 *vi.*重申，重复强调；重新开始
diverge [dɪ'vɜ:dʒ] *vi.* 分开，叉开；偏离；[数]发散，无极限 *vt.* 使发散；使偏离
homogeneity [ˌhɔməudʒe'ni:iti] *n.* 同质，同次性；齐性；均匀性
sensor ['sensə:] *n.* 传感器，灵敏元件
node [nəud] *n.* 节点；（计算机网络的）节点
parameter [pə'ræmitə] *n.* [数]参数；[物][数]参量；限制因素；决定因素
concentration[ˌkɔnsən'treiʃən] *n.* 集中；专心；关注；浓度
GUI：graphical user interface 图形用户界面

9.2 Reading Materials

1．'Internet of Things' Joint Research

INTEL DEVELOPER FORUM, Beijing, April 11, 2012 – Intel Corporation announced today a joint agreement with the Beijing Municipal Government and Institute of Automation of Chinese Academy of Sciences to establish "China Intel Internet of Things" (IoT) Joint Labs to collaborate on IoT-related core technology research, development and business model innovation. The three organizations will invest 200 million RMB (US$31.7 million) over the next 5 years.

IoT is a global technology evolution through which data from billions of devices are seamlessly connected, intelligently managed and securely interacting over a network. This enables people, devices and systems to turn data into useful information and valued services.

The foundation of the joint research will address the core technologies associated with IoT including sensing, networking and processing, among others that will help address computing, storage and communication challenges of massive-scale systems derived from the large amounts of data gathered.

Tan Tieniu, director of National Laboratory of Pattern Recognition Institute of Automation of Chinese Academy of Sciences, and Jesse Fang, vice president of Intel Labs and managing director of Intel Labs China, were named co-presidents of the China Intel IoT Joint Labs.

"The China Intel IoT joint labs is the largest of its kind that Intel has participated in research collaboration in China, and it is unique in that Intel not only contributed funding but also employees as dedicated senior research leaders,"said Fang. "New applications from personalized energy management to smart traffic control to smart cities will be made possible by the research conducted at these labs. "

2．What is a Smart Building?

The phrase "smart building" conjures up images of sleek new structures incorporating all the latest energy-saving bells and whistles, but that's only part of the story. The fact is, any number of "smart" elements can apply to older, existing buildings. In addition to cutting energy costs,

upgrading an older building can also result in a more comfortable and healthy environment for employees and customers, it can contribute to the quality of life in its community, and it can provide businesses with a green marketing tool that boosts their public profile.

The challenge for property owners is to find the right type and combination of smart elements that provide them with the greatest return on their investment.

Because there are so many older, inefficient buildings in the U.S., the potential for energy savings goes beyond benefits to individual property owners. It's a matter of national energy policy, too, and the Department of Energy (DOE) has a long history of working with the building industry and related sectors to make improvements.

One good example, is the industry's response to stepped-up efficiency standards for HVACR (heat, ventilation, air conditioning and refrigeration systems), which lead to the development and widespread use of moreadvanced compressors.

As for the numbers, according to DOE there are about 4.8 million commercial buildings and 350,000 industrial facilities in the U.S., which together account for about half of the country's total energy use.

The annual energy costs for those buildings add up to a total of about $202.3 billion, and DOE estimates that a good 30 percent of that energy is used "inefficiently or unnecessarily."

In other words, despite past improvements there is still plenty of wiggle room to make older buildings operate more efficiently and save money for their owners and tenants.

9.3 Exercises

1．Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）物联网
（2）射频身份识别码
（3）电子标签
（4）条形码
（5）二维码
（6）智慧城市
（7）信息通信技术
（8）无线传感器网络
（9）分布式网络
（10）自动控制
（11）图形用户界面

2．ead the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）The Internet of Things refers to uniquely（　　）objects (things) and their virtual representations in an（　　）structure.

（2）（　　） is often seen as a prerequisite for the Internet of Things.

（3）If all objects and people in daily life were equipped with （　　）, they could be identified and inventoried by computers. However, unique identification of things may be achieved through other means such as（　　）or（　　）as well.

（4）Equipping all objects in the world with （　　　）devices could be transformative of daily life.

（5）Alcatel-Lucent touchatag service and Violet's Mirror gadget provide a pragmatic consumer oriented approach to the Internet of Things by which a developer can link real world items to the online world using（　　）tags and （　　）.

（6）It is against this background that the concept of the smart city has been introduced as a （　　）device to encompass modern urban production factors in a common framework and to highlight the growing importance of （　　）, social and environmental capital in profiling the competitiveness of cities.

（7）The significance of these two assets - social and environmental capital - itself goes a long way to（　　）smart cities from their more technology-laden counterparts, drawing a clear line between them and what goes under the name of either digital or（　　）cities.

（8）The data is delivered（　　）and in（　　）to the citizens or the appropriate authorities.

（9）From the moment hotel guests arrive, their stay can be（　　）to meet their personal preferences.

（10）Guests can request services from the bellman's desk, valet, spa, golf club, front desk and more—right from a（　　）or the TV in their room. Hoteliers can choose from a wide range of services to offer your guests, while making（　　）functions like mini-bar refills, housekeeping services and security settings more efficient to manage.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（　　）The Internet of Things (IoT) refers to uniquely identifiable objects (things) and their virtual representations in an Internet-like structure.

（2）（　　）Radio-frequency identification (RFID) is often seen as a prerequisite for the Internet of Things. If all objects and people in daily life were equipped with radio tags, they could be identified and inventoried by computers. So that unique identification of things can be only achieved through RFID.

（3）（　　）An approach taken by the Media and Graphics lab at the University of British Columbia (Canada) focuses on a lightweight toolkit for developing IoT applications and targets rapid development using Web technologies and protocols.

（4）（　　）Its main focus is still on the role of ICT infrastructure, but much research has also been carried out on the role of human capital/education, social and relational capital and environmental interest as important drivers of urban growth.

（5）（　　）Wired sensor networks is a specific technology that helps to create Smart Cities. The aim is to create a distributed network of intelligent sensor nodes which can measure many parameters for a more efficient management of the city.

（6）（　　）Vehicle traffic can be monitored in order to modify the city lights in a static way.

（7）（　　）The availability and quality of the ICT infrastructure is the only definition of a smart or intelligent city.

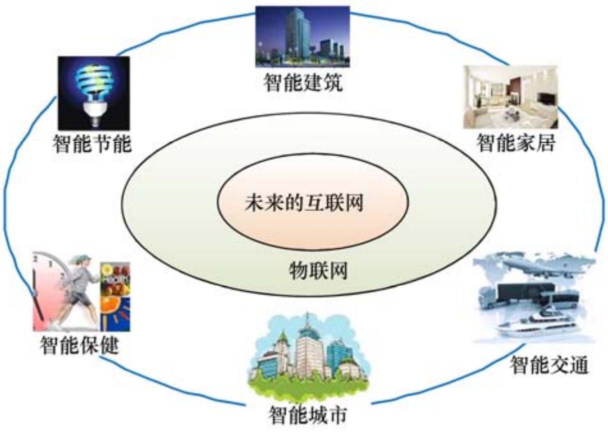
（8）（　　）At a mesoregional level, we observe renewed attention for the role of soft communication infrastructure in determining economic performance.

（9）（　　）Pre-set “scenes” make it easy for guests to match their mood or preference in one touch.

9.4　课文参考译文　物联网

9.4.1　物联网概述

物联网（IoT）类似因特网，是由一个个节点组成的架构，每一个节点代表一个独一无二的对象（物体）。1999 年，Adam Baumgarten 首先提出了物联网这个词。而这个词为公众所熟知则是由于 Auo-ID 实验室和相关的市场调研报告。射频身份识别码（RFID，通常也翻译为电子标签）通常被认为是物联网得以实现的前提。如果日常生活中所有的对象和人都配备一个电子标签，它们都将能够被计算机收录和识别。不过，也可以采用条形码和二维码等其他方法来唯一确定一件物体。



译图 9-1　物联网与智慧地球

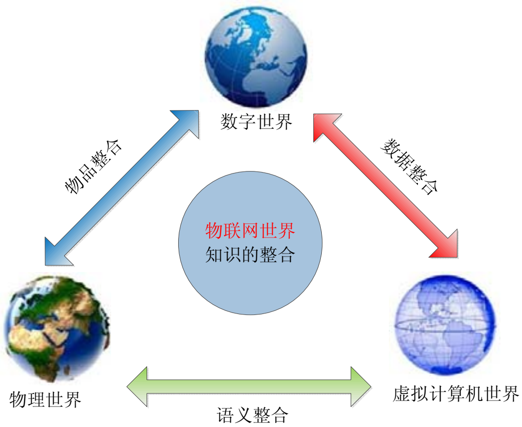
给这个世界上的所有物体都配备一个微型识别装置将会颠覆我们的日常生活方式。例如，工商企业再也不会缺货，也不会产出无用的产品，因为交易各方都知道市场需要什么。按照当前的终端用户协议，一个人同物品的互动能力将会基于及时或目前的需求而改变。

物联网的应用领域包括废弃物管理，城市规划，可持续发展，不间断护理，应急响应，智能购物，智能产品管理，智能测量，家庭自动化和其他智能事件等。阿尔卡特-朗讯的 touchatag 服务和 Vilet 公司的迷你装置 Mirror 提供了一种面向消费者的实用的手段去接入物联网。有了这种手段，开发者就可以利用电子标签和 QR 码把现实世界和线上世界连接在一起。

物联网面临的一个关键问题是如何快速地开发应用。加拿大的 British Columbia 大学正集中精力研发一个轻量级的工具包，目标是通过这个工具包，可以利用 Web 技术和协议，快速地开发物联网应用。该工具包建立在前期的物联网研究基础上，特别是发表于 IoT 2010（IEEE）大会上的 MAGIC Broker 技术。在 IoT 2012（IEEE）大会上，已经对该工具包进行了阐述。

9.4.2 智慧城市

当今城市的好坏不仅决定于城市的硬件基础设施（物质资本），而且越来越多的决定于知识传播效率和社会化基础设施（智力资本和社会资本）。后一种资本对于城市的竞争力更加起着决定性的作用。正是在这种背景下，提出了智慧城市的概念。从战略的角度来看，这一概念把组成现代城市的要素融合在了一个统一的框架里面，并且突出了信息通信技术（ICTs）以及社会和环境资本在代表城市的竞争力方面日益增长的重要性。人们花了很长时间才把智慧城市和数字城市，或者说智能城市，区分开来，后者更强调现代技术的重要性。



译图 9-2　物联网——物理世界、数字世界与虚拟社会之间的交互关系

近些年，随着下一波城市化进程的到来，智慧城市的概念在政策方面变得非常时髦，用智慧城市这个词，主要是为了和数字城市或者说智能城市区别开来。虽然智慧城市仍旧侧重于信息通信基础设施的建设，但是对于人力资本和教育，社会和关联资本以及环境因素对城市增长的驱动作用，人们已经展开了大量的研究。

特别是欧盟，一直在努力地设计一套政策， 以促进那些大城市“智慧地”增长。其他的一些国际机构和智库也认可这种以有线 ICT（wired ICT）为驱动力的发展模式。例如，智能社区论坛致力于研究全局 ICT 革命的局部效应。The OECD and EUROSTAT Oslo Manual 则强调 ICT 革命本身的作用，他们还提供了一个工具包，以便研究人员能够以一套成熟的，统一的标准测定某些指标。在城域级别上，我们更加重视通信软设施对经济数据的影响。

ICT 设施的便利性和通信质量并不是一个智慧城市的全部，人力资源和教育培训也起着很重要的作用。例如，过去的经验表明，如果一个城市的劳动力受教育程度比较高，那么这个城市发展得也较为迅速。

变革来自于这样的一些企业家：他们所在的行业和他们的产品对于劳动力的技能要求越来越高。在人力资源的投资方面，并不是所有的城市都取得了同样的成功，受过良好教育的劳动力（创意阶层）正逐渐地聚集在一起。具有人才多样化的城市已经引起了研究者和政策制定者的注意。现在已经表明，那些在过去就拥有熟练技工的城市，现在吸引到了更多的技

工，而他的对手们则在这场竞争中失败了。政策制定者们，尤其是欧洲的政策制定者，非常重视各个地区的均衡发展。如此看来，人才的不断集聚反倒成为了一个大问题。

无线传感器网是一项有助于创建智慧城市的特别技术。这项技术的目标是创建一个分布式的智能传感器网络，这些传感器能够采集许多参数，以促进更加高效地管理城市。这些采集到的数据以无线的方式 实时地传送给居民或相关机构。

例如，居民能够检测到每一条街道的污染浓度，或者当放射水平升高到某一值时，他们也可以收到自动的警报。有关当局还可以优化公园的浇灌和城市的照明。水管泄漏点和噪音分布也可以很容易地获取到。当垃圾箱快要装满时，它可以发出一个警报。

为了动态的控制交通信号，可以监测机动车流量。通过侦测最近的停车场在哪里，交通量可以得到减少。司机能够得到及时的信息，以便他们能够快速地找到免费停车场，节省时间和燃油。这些信息能够减少交通拥堵，提高生活质量。

9.4.3　物联网的应用——Control 4 酒店解决方案

从门到窗帘，Control 4 酒店解决方案能够真正给客人带来“惊叹的体验”。从客人到达酒店的那一刻起，他们的起居就可以根据个人喜好进行设置。Control 4 可能实现哪些功能？让我们来举几个例子：

特别欢迎

当客人要进入自己的房间或套房时，系统会识别客人是否第一次到来，并对他们的到来表示欢迎。于是，灯光照亮整个房间，窗帘自动打开展示出美丽的景色，电视机显示的列表为客人提供个性化的自动控制。

完全遥控

电视作为一个通讯中心，可以显示新的语音邮件，包裹递送等消息。客人还可以通过电视的图形用户界面控制音乐，灯光，温控器，壁炉，窗帘，免打扰服务，以及打扫房间等酒店服务。

一触式服务

客人可以通过触摸屏或客房中的电视要求行李寄存中心、代客泊车、水疗馆、高尔夫俱乐部、前台等提供服务。同时，酒店管理者也可以选择为客人提供一系列的服务，如迷你酒吧，客房服务和安全设置等，使得后台的管理更加高效。

个性化场景

轻触一下，客人即可以根据自己的心情或喜好，进入预先设定的“场景”。例如，床边的“晚安”按钮可以关闭灯，电视和/或音乐；关上窗帘，开启房间的免打扰模式。同样，客人可以个性化其唤醒情境——轻柔的音乐或早间新闻，灯光渐渐地亮起，房间达到理想的温度。当一切就绪，打开窗帘，迎接早晨的太阳。

轻松“绿化”

可以请客人参与到酒店首选的绿色设定上。通过一个简单的“绿色按钮”，自动管理床单和毛巾服务，温度和照明，实现节能最大化。

Unit 10　4G Overview

10.1　Text

10.1.1　Introduction of 4G

While most of us are used to getting high speed Internet connections at home, the office or even the local coffee shop, once we are on the road those high speeds have to stay behind. With 4G the promise is that you can get real mobile broadband to go. In this piece we are going to tell you all about the technology and its benefits, who offers it or plans to, how much it costs, and the gear you need to enjoy the next generation of wireless broadband today.

But first, some background: 4G is the short name for fourth-generation wireless, the stage of mobile communications that will enable things like IP-based voice, data, gaming services and high quality streamed multimedia on portable devices with cable modem-like transmission speeds. It's a successor to 2G and 3G wireless, whereby the first signified the shift from analog to digital transmissions, bringing data services like SMS and email to mobile phones for the first time, and the second refers to the advent of things like global roaming as well as higher data rates.

Think of wireless generations as a handful of services that get faster and more feature-rich as newer technology becomes available. The 3G networks that we use today allow us to stream video, download music and files, and surf the web at average download speeds from 600Kbps to 1.4Mbps. With 4G you'll be able to do the same but at much faster rates, while the extra bandwidth opens the door for newer applications. The Download and upload date rates of cellular systems are shown in Table 10-1.

There are a number of standards and technologies pertaining to each wireless generation — GSM, cdmaOne, GPRS, EDGE, CDMA2000, UMTS (also marketed as 3GSM), HSDPA, among others. For practical reasons, we won't be dwelling on the technicalities of each term and instead will move onto the ones that involve our topic of interest here: 4G.

Although no set of standards have been established as of yet by the International Telecommunication Union (ITU), the authority on such things, two competing technologies have been proposed: LTE and WiMAX. Many service providers often use the term 4G mobile broadband to describe the technologies they are offering based on their own, sometimes distorted definitions. However, current implementations are largely considered pre-4G, as they don't fully comply with the planned requirements of 1Gbps for stationary reception and 100Mbps for mobile.

Table 10-1　Download/Upload Data rates of Cellular Systems

		Real World(avg)		Theoretical(max)		Availability
		Download	Upload	Download	Upload	
2.5G	GPRS	32～48Kbps	15Kbps	114Kbps	20 Kbps	Today
2.75G	EDGE	175 Kbps	30 Kbps	384 Kbps	60 Kbps	Today

续表

		Real World(avg)		Theoretical(max)		Availability
		Download	Upload	Download	Upload	
3G	UMTS	226 Kbps	30 Kbps	384 Kbps	64 Kbps	Today
	W-CDMA	800 Kbps	60 Kbps	2 Mbps	153 Kbps	Today
	EV-DO Rev. A	1 Mbps	500 Kbps	3.1 Mbps	1.8 Mbps	Today
	HSPA 3.6	650 Kbps	260 Kbps	3.6 Mbps	384 Kbps	Today
	HSPA 7.2	1.4 Mbps	700 Kbps	7.2 Mbps	2 Mbps	Today
Pre-4G	WiMAX	3～6 Mbps	1 Mbps	100 Mbps+	56 Mbps	Today
	LTE	5～12 Mbps	2～5 Mbps	100 Mbps+	50 Mbps	End 2010
	HSPA+	–	–	56 Mbps	22 Mbps	2011
	HSPA 14	2 Mbps	700 Kbps	14 Mbps	5.7 Mbps	Today*
4G	WiMAX 2 (802.16m)	–	–	100 Mbps mobile/ 1 Gbps fixed	60 Mbps	2012
	LTE Advanced	–	–	100 Mbps mobile/ 1 Gbps fixed	–	2012+

Besides speed, several other guidelines have been traced for wireless communication standards to qualify as 4G. In a nutshell, they should be very spectrally efficient, should dynamically share and utilize the network resources to support more simultaneous users per cell, have smooth hangovers across heterogeneous networks, offer high quality of service for next generation multimedia support, and should be based on an all-IP packet switched network.

10.1.2 Evolution from 1G to 4G

New mobile generations have appeared about every ten years since the first move from 1981 analog (1G) to digital (2G) transmission in 1992. This was followed, in 2001, by 3G multi-media support, spread spectrum transmission and at least 200 Kbps peak bit rate, in 2011/2012 expected to be followed by “real” 4G, which refers to all-Internet Protocol (IP) packet-switched networks giving Ultra Mobile Broadband (gigabit speed) access.

While the ITU has adopted recommendations for technologies that would be used for future global communications, they do not actually perform the standardization or development work themselves, instead relying on the work of other standards bodies such as IEEE, The WiMAX Forum and 3GPP.

In mid-1990s, the ITU-R standardization organization released the IMT-2000 requirements as a framework for what standards should be considered 3G systems, requiring 200 Kbps peak bit rate. In 2008, ITU-R specified the IMT-Advanced (International Mobile Telecommunica-tions Advanced) requirements for 4G systems.

The fastest 3G-based standard in the UMTS family is the HSPA+ standard, which is

commercially available since 2009 and offers 28 Mbps downstream (22 Mbps upstream) without MIMO, i.e. only with one antenna, and in 2011 accelerated up to 42 Mbps peak bit rate downstream using either DC-HSPA+ (simultaneous use of two 5 MHz UMTS carrier) or 2×2 MIMO. In theory speeds up to 672 Mbps is possible, but has not been deployed yet. The fastest 3G-based standard in the CDMA2000 family is the EV-DO Rev. B, which is available since 2010 and offers 15.67 Mbps downstream.

10.1.3 LTE

Short for Long-Term Evolution, LTE is considered by many to be the natural successor to current-generation 3G technologies, in part because it updates UMTS networks to provide significantly faster data rates for both uploading and downloading. The specification calls for downlink peak rates of at least 100Mbps and an uplink of 50Mbps, but going by real world tests its transfer speeds will more likely range from 5～12Mbps for downloads and 2～5Mbps for uploads.

LTE is being developed by a group of telecommunications associations known as the 3rd Generation Partnership Project, or 3GPP, as an eight release of what has been evolving since 1992 from the GSM family of standards. The logo of LTE is shown in Figure 10-1.

Figure 10-1 Logo of LTE

There are two fundamental aspects of LTE. The first is that the technology finally leaves behind the circuit switched network of its GSM roots and moves to an all-IP flat networking architecture. This is a significant shift which in very simple terms means that LTE will treat everything it transmits, even voice, as data. The other big change relates to the use of MIMO technology, or multiple antennas at both the transmitter and receiver end to improve communication performance. This setup can either be used to increase the throughput data rates or to reduce interference.

Many big-name global operators and mobile communications companies are backing LTE in the race for 4G mobile broadband, including Vodafone, Orange, T-Mobile, LG Electronics, Ericsson, Nokia, Siemens, NTT DoMoCo, and others. In the U.S., Verizon Wireless has said it is going commercial with its LTE network in the fourth quarter, with 25～30 markets up and ready at launch. AT&T and T-Mobile claim they will begin to deploy LTE in 2011, but in the meantime both networks have moved to HSPA 7.2 and the latter plans to roll out HSPA+beginning this year. Theoretically these can support speeds of up to 7.2 and 21 Mbps, respectively, but in real world scenarios they are only marginally faster than most 3G data services.

The reason behind LTE’s strong industry support lies in the relative ease of upgrading from current 3G networks worldwide over to LTE mobile broadband, compared to the significant infrastructure build out that WiMAX has taken thus far. Fewer cell sites have to be built and penetration into buildings is better at the 700 MHz spectrum LTE uses. However, WiMAX deployments are already up and running while LTE's formal debut is still a few months out.

10.1.4 WiMAX

WiMAX is a wireless broadband access standard developed and maintained by the IEEE under the 802.16 designation. As its name suggest, WiMAX can be thought of as an extension of Wi-Fi designed to enable pervasive, high-speed mobile Internet access on a wide range of devices, from laptops to smartphones. The current implementation is based on the 802.16e specification which offers theoretical downlink rates upwards of 70Mbps and up to 30-mile ranges.

Again, “theoretical” is the keyword here as WiMAX, like all wireless technologies, can either operate at higher bitrates or over longer distances but not both. Production networks being operated in the United States are seeing average speeds go from 3 to 6Mbps, with bursts up to 10Mbps. Like LTE — and Wi-Fi 802.11n for that matter — WiMAX supports MIMO technology, which means that additional antennas can increase the potential throughput. The Logo of WiMAX is shown in Figure 10-2.

Figure 10-2 Logo of WiMAX

There is no uniform global licensed spectrum for WiMAX, but three have been listed: 2.3 GHz, 2.5 GHz and 3.5 GHz. In the U.S., the biggest segment available is around 2.5 GHz and is already assigned primarily to Clearwire, a wireless internet service provider in which Sprint Nextel holds a majority stake.

In terms of total available 4G spectrum to deploy their services, Clearwire has several times more than its competitors, which have smaller portions of the 700 MHz band. However, Verizon and AT&T are not too worried about this as they can re-utilize spectrum being used right now for 2G and 3G services by upgrading these to LTE when the demand is there.

Furthermore, as mentioned earlier, the 700 MHz band that both Verizon and AT&T plan to use has enormously better range and penetration of buildings than the same power of signal at 2.5 GHz. Some experts have said that 700MHz will require as few as one-quarter as many base stations to offer identical coverage to 2.5 GHz.

As you might have guessed, the industry players behind these 4G technologies reflect the history of each standard. Whereas LTE biggest supporters are, in general, telecommunication service companies and handset manufacturers, WiMAX counts the likes of Intel, Cisco and Google among its most important backers. It should be noted though that many companies like Nokia or Motorola are members of both industry groups, with different levels of involvement.

Technical words and phrases

streamed multimedia　流媒体

SMS（Short Messaging Service）　短消息服务

roam　[rəum]　vt.& *vi*.　漫游

pertain　[pə'tein]　*vi*.　关于，有关；适合；附属，从属

EDGE （Enhanced Data Rate for GSM Evolution）　增强型数据速率 GSM 演进技术

UMTS（Universal Mobile Telecommunications System）　通用移动通信系统

HSDPA（High Speed Downlink Packet Access） 高速下行分组接入
dwell [dwel] *vi.* 居住；存在于；细想某事
technicality [ˌtekniˈkæləti] *n.* 专门性，学术性，技术性；术语；专门事项
ITU（International Telecommunication Union） 国际电信联盟
LTE（Long-Term Evolution） 长期演进
WiMAX（Worldwide Interoperability for Microwave Access） 全球微波互联接入
nutshell [ˈnʌtʃel] *n.* 简言之，一言以蔽之
handover [ˈhændəuvə(r)] *n.* 移交；交接
heterogeneous [ˌhetərəˈdʒiːniəs] *adj.* 多种多样的；非均匀；错杂
packet switched network 分组交换网络
spread spectrum transmission 扩频通信
gigabit [ˈdʒigəbit] 千兆比特
IEEE（Institute of Electrical and Electronics Engineers） 美国电气和电子工程师协会
3GPP （The 3rd Generation Partnership Project） 第三代合作伙伴计划
MIMO（Multi-input Multi-output） 多入多出技术
throughput [ˈθruːput] *n.* 生产量，生产能力，吞吐量；流率
marginally [ˈmɑːdʒinəli] *adv.* 略微；稍微
infrastructure [ˈɪnfrəstrʌktʃə(r)] *n.* 基础设施；基础建设
penetration [ˌpeniˈtreiʃn] *n.* 渗透；穿透
laptops [ˈlæptɒp] *n.* 便携式电脑
smartphone 智能手机
upgrade [ˌʌpˈgreid] *vt.* 提升；使（机器、计算机系统等）升级
handset [ˈhændset] *n.* 电话听筒；手机；手持机

10.2 Reading Materials

1. Wi-Fi

Wi-Fi is a popular technology that allows an electronic device to exchange data wirelessly (using radio waves) over a computer network, including high-speed Internet connections. The Wi-Fi Alliance defines Wi-Fi as any “wireless local area network (WLAN) products that are based on the Institute of Electrical and Electronics Engineers' (IEEE) 802.11 standards”. However, since most modern WLANs are based on these standards, the term “Wi-Fi” is used in general English as a synonym for “WLAN”.

A device using Wi-Fi, such as a personal computer, video game console, smartphone, tablet, or digital audio player, can connect to a network resource such as the Internet via a wireless network access point. Such an access point (or hotspot) has a range of about 20 meters (65 ft) indoors and a greater range outdoors. Hotspot coverage can comprise an area as small as a single room with walls that block radio signals or a large area up to many square miles, which can be covered by multiple overlapping access points.

2. Bluetooth

Bluetooth is a proprietary open wireless technology standard for exchanging data over short distances (using short-wavelength radio transmissions in the ISM band from 2400～2480 MHz) from fixed and mobile devices, creating personal area networks (PANs) with high levels of security. Created by telecoms vendor Ericsson in 1994, it was originally conceived as a wireless alternative to RS-232 data cables. It can connect several devices, overcoming problems of synchronization.

Bluetooth is managed by the Bluetooth Special Interest Group, which has more than 15 000 member companies in the areas of telecommunication, computing, networking, and consumer electronics. The SIG oversees the development of the specification, manages the qualification program, and protects the trademarks. To be marketed as a Bluetooth device, it must be qualified to standards defined by the SIG. A network of patents is required to implement the technology and are only licensed to those qualifying devices; thus the protocol, whilst open, may be regarded as proprietary.

3. Cloud computing

Cloud computing is the use of computing resources (hardware and software) that are delivered as a service over a network (typically the Internet). The name comes from the use of a cloud-shaped symbol as an abstraction for the complex infrastructure it contains in system diagrams. Cloud computing entrusts remote services with a user's data, software and computation.

In the business model using software as a service, users are provided access to application software and databases. The cloud providers manage the infrastructure and platforms on which the applications run. SaaS is sometimes referred to as “on-demand software” and is usually priced on a pay-per-use basis.

Proponents claim that the SaaS allows a business the potential to reduce IT operational costs by outsourcing hardware and software maintenance and support to the cloud provider. This enables the business to reallocate IT operations costs away from hardware/software spending and personnel expenses, towards meeting other IT goals. In addition, with applications hosted centrally, updates can be released without the need for users to install new software. One drawback of SaaS is that the users' data are stored on the cloud provider's server. As a result, there could be unauthorized access to the data.

10.3 Exercises

1．Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）流媒体
（2）全球漫游
（3）高速下行分组接入
（4）分组交换网络

（5）电路交换网络
（6）扩频通信
（7）多入多出技术
（8）基站
（9）授权频段
（10）多天线系统

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）4G is the short name for（　　） wireless, the stage of mobile communications that will enable things like （　　）-based voice, data, gaming services and high quality （　　） multimedia on portable devices with cable modem-like transmission speeds.

（2）Although no set of standards have been established as of yet by the International Telecommunication Union, the authority on such things, two competing technologies have been proposed:（　　） and （　　）.

（3）Wireless communication standards should be very （　　） efficient, should （　　） share and utilize the network resources to support more simultaneous users per （　　）.

（4）The ITU-R standardization organization released the (　　) requirements as a framework for what standards should be considered 3G systems, requiring 200 kbps (　　) bit rate.

（5）The fastest 3G-based standard in the UMTS family is the （　　） standard.

（6）Short for Long-Term Evolution, （　　） is considered by many to be the natural successor to current-generation 3G technologies

（7）LTE is being developed by a group of telecommunications associations known as the 3rd Generation Partnership Project, or （　　）, as an eight （　　） of what has been evolving since 1992 from the GSM family of standards.

（8）There are two fundamental aspects of LTE. The first is that the technology finally leaves behind the（　　）switched network of its GSM roots and moves to an （　　）flat networking architecture.

（9）The other big change relates to the use of MIMO technology, or multiple antennas at both the （　　）and（　　）end to improve communication performance.

（10）WiMAX can be thought of as an extension of（　　）designed to enable pervasive, high-speed mobile Internet（　　）on a wide range of devices.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（　　）4G will enable things like IP-based voice and high quality streamed multimedia on portable devices with cable modem-like transmission speeds.

（2）（　　）With 4G you'll be able surf the web at average download speeds from 600Kbps to 1.4Mbps.

（3）（　　）Two competing technologies which have been proposed to 4G are LTE and Wi-Fi.

（4）（　　）The planned requirements of 4G are 1Gbps for stationary reception and 100Mbps for mobile.
（5）（　　）Real 4G refers to all-Internet Protocol (IP) packet-switched networks.
（6）（　　）ITU performs the standardization or development work themselves.
（7）（　　）In 2008, ITU-R specified the IMT-Advanced requirements for 4G systems.
（8）（　　）LTE is based on an all-IP flat networking architecture.
（9）（　　）LTE uses single antenna at both the transmitter and receiver end to improve communication performance.
（10）（　　）WiMAX is a wireless broadband access standard developed and maintained by ITU.
（11）（　　）There is no uniform global licensed spectrum for WiMAX.

10.4　课文参考译文　4G 概述

10.4.1　4G 介绍

大多数人习惯在家里、办公室或者当地的咖啡店获得高速互联网连接，一旦我们处于行进状态，那些高速网络只能空置在那里。4G 承诺你可以带着真正的移动宽带上路。本文将向你讲述有关 4G 技术的相关情况和它的益处，谁提供或计划提供此项技术，它价值几何，以及享受下一代无线宽带所需的设备。

但是首先，相关背景如下：4G 是第四代无线通信的简称，4G 是移动通信发展的一个阶段，支持便携式设备以电缆调制解调器的传输速度传输基于 IP 的语音、数据、游戏业务和高质量的流媒体。4G 是对 2G 和 3G 的继承，据此可知，首先，4G 意味着从模拟传输到数字传输的转变，第一次将诸如短信群发业务和电子邮件等数据业务融入到移动电话；其次，4G 预示了全球漫游以及高数据传输速率的到来。

几个新技术中，无线通信被认为是少数几个服务中能获得更快更丰富的功能服务。现今所使用的 3G 网络，让我们以 600Kbps～1.4Mbps 的速度观看视频流，下载音乐文件以及上网冲浪。拥有 4G 后，你将能以更快的速度做同样的事情，但与此同时，你也将为新的应用付出额外的带宽费用。蜂窝移动通信系统的数据上传/下载速率如下表 10-1 所示。

目前有许多标准和技术对应不同的无线通信制式——GSM，cdmaOne，GPRS，EDGE，CDMA2000，UMTS（也被市场称为 3GSM），HSDPA 等等。出于实际原因，我们不希望过多关注以上各种通信制式，而直接进入我们感兴趣的主题：4G。

尽管国际电信联盟（ITU）还未确立一套标准，但已出现了两种颇有权威性和竞争力的技术：LTE 和 WiMAX。许多业务提供商根据自己的理解将他们提供的技术用 4G 移动宽带这一术语来进行描述，但事实上却时常歪曲了这一技术的定义。现在的实施效果没有完全满足原计划要求的静止状态 1Gbps、移动状态 100Mbps 的数据传输速率，因此只能被认为是准 4G。

除了速度，为了使无线通信标准符合 4G 的要求，必须探索另外几个指导性原则。简而言之，这些标准应该具有一些基本特征，包括：频谱利用率高、动态共享，可利用网络资源支持更多同时来自单个蜂窝的用户数量，能够从多种网络平稳过渡，为下一代多媒体提供高质量的服务，以及应基于 IP 分组交换网络。

译表 10-1　蜂窝移动通信系统数据上传/下载速率

		实际速率(平均)		理论速率(最大)		应用
		下载	上传	下载	上传	
2.5G	GPRS	32-48Kbps	15Kbps	114Kbps	20 Kbps	当前
2.75G	EDGE	175 Kbps	30 Kbps	384 Kbps	60 Kbps	当前
3G	UMTS	226 Kbps	30 Kbps	384 Kbps	64 Kbps	当前
	W-CDMA	800 Kbps	60 Kbps	2Mbps	153 Kbps	当前
	EV-DO Rev. A	1Mbps	500 Kbps	3.1Mbps	1.8Mbps	当前
	HSPA 3.6	650 Kbps	260 Kbps	3.6Mbps	384 Kbps	当前
	HSPA 7.2	1.4Mbps	700 Kbps	7.2Mbps	2Mbps	当前
前 4G	WiMAX	3～6Mbps	1 Mbps	100 Mbps+	56 Mbps	当前
	LTE	5～12 Mbps	2～5 Mbps	100 Mbps+	50 Mbps	2010 年底
	HSPA+	—	—	56 Mbps	22 Mbps	2011
	HSPA 14	2 Mbps	700Kbps	14 Mbps	5.7 Mbps	当前*
4G	WiMAX 2 (802.16m)	—	—	100Mbps mobile/ 1Gbps fixed	60 Mbps	2012
	LTE 进阶版	—	—	100Mbps mobile/ 1Gbps fixed	—	2012+

10.4.2　1G 到 4G 的演进

1992 年，移动通信第一次从 1981 年开始的模拟传输（1G）转变到数字传输（2G），自此新的移动时代每 10 年更新一次，在 2001 年演进到 3G，可支持多媒体数据传输、扩频通信以及最高 200Kbps 的数据传输速率，并在 2011—2012 年演进到基于互联网协议（IP）分组交换网络的真 4G 时代，提供超宽带的移动接入（千兆速度）。

虽然国际电信联盟采纳了用于未来全球通信的技术建议，但实际上他们并不自行完成标准制定和规划发展，而是依靠其他标准化机构来完成，如 IEEE、WiMAX 论坛和 3GPP。

20 世纪 90 年代中期，ITU-R 标准化组织发布了 IMT-2000 的框架要求，明确了最高数据传输率达 200Kbps 的 3G 系统应采用何种标准。2008 年，ITU-R 提出了用于 4G 系统的 IMT-Advanced（高级国际移动通信）要求。

HSPA+标准是 UMTS 家族中最快的 3G 标准，于 2009 年投入商用，在没有使用 MIMO 技术（即只使用单个天线）的前提下能够提供 28 Mbps 的下行速度（22 Mbps 的上行速度），2011 年，当任意使用 DC-HSPA+（同步使用两个 5MHz UMTS 载波）或 2×2MIMO 中的一种技术，其上行峰值数据传输速度可提高到 42Mbps。理论上高达 672Mbps 的速度是可行的，但现在还没有实现。 EV-DO Rev. B 是 CDMA2000 家族中最快的 3G 标准，于 2010 年投入使用，提供 15.67Mbps 的下行速度。

10.4.3 LTE

LTE 是长期演进的简称，LTE 被许多人认为是当前阶段 3G 技术的接班人，某种程度上是因为它更新了 UMTS 网络，提供更快的上传和下载速率。规范要求峰值下行速率至少 100Mbps，上行速率至少 50Mbps，但实测数据表明，其传输速度范围更接近下行 5～12Mbps，上行 2～5Mbps。

LTE 是电信协会第三代合作伙伴计划（3GPP）开发的，从 1992 年 GSM 标准至今共经历了 8 个版本的演进。LTE 图标如下图 10-1 所示。

译图 10-1 LTE 图标

LTE 有两个基本方面。第一，该技术最终放弃了 GSM 采用的电路交换网络转而采用全 IP 扁平网络架构。这是一个重大的转变，简言之，这种转变意味着 LTE 将传输的所有东西，包括声音，看成是数据。LTE 的另一个重大改变则是通过使用 MIMO 技术，在发射、接收端使用多天线系统来来增加数据吞吐率或减小干扰，以提高通信质量。

多家全球知名的运营商和移动通信公司，如 Vodafone、Orange、T-Mobile、LG Electronics、Ericsson、Nokia、Siemens、NTT DoMoCo 等，都在 4G 移动宽带的竞争中支持 LTE。在美国，Verizon 无线公司声称准备在第四季度将其 LTE 网络投入商用，使市场占有率达 25%～30%，目前已准备就绪。AT&T 和 T-Mobile 公司则声称将在 2011 年展开 LTE 业务，但与此同时，两个网络目前处于 HSPA 7.2 阶段，后者计划今年初不再沿用 HSPA+。理论上 HSPA7.2 和 HSPA+支持的下行速度分别可达 7.2Mbps 和 21Mbps，但实际上仅仅略快于大部分 3G 网络的数据传输速率。

LTE 背后强有力的产业支持原因在于，相比迄今为止 WiMAX 已建成的重要基础设施，从现行的全球 3G 网络升级到 LTE 移动宽带相对容易。LTE 需要建设的蜂窝基站较少，所用的 700MHz 频段信号穿透建筑物的能力较强。然而，WiMAX 的部署早已启动并运行，而 LTE 的正式登场却只有几个月。

10.4.4 WiMAX

WiMAX 是基于 IEEE 802.16 标准发展而来的宽带无线接入标准。正如其名字所示，WiMAX 被认为是 Wi-Fi 的延伸，能将便携式电脑、智能手机等多种终端设备以大范围、高速移动的方式接入互联网。目前，WiMAX 的实现基于 802.16e 标准，其理论下行速度超过 70Mbps，传输范围超过 30m。

值得注意的是这里的“理论”一词，和所有的无线技术一样，WiMAX 可以工作于更高的传输速率或更长的距离，但不能两者同时具有。在美国，现行 WiMAX 网络平均速度 3～6Mbps，对突发信号可达 10Mbps。WiMAX 同 LTE 和 Wi-Fi 802.11n 一样支持 MIMO 技术，这意味着 WiMAX 可通过增加额外的天线来达到潜在数据吞吐量的增加。WiMAX 图标如图 10-2 所示。

译图 10-2 WiMAX 图标

WiMAX 无需授权频段，这里仅列出其中三个：2.3GHz，2.5GHz 和 3.5GHz。在美国，使用最多的是 2.5GHz 频段，这一频段主要分配给了一家由美国第三大移动运营商 Sprint Nexte 控股的无线网络业务供应商

Clearwire。

就可用于开展 4G 业务的频段而言，Clearwire 拥有的频段资源是其竞争对手的好几倍，而其竞争对手只分配到 700MHz 段的小部分。然而，Verizon and AT&T 公司对此无需过于担心，因为如果有需要，他们可以通过将 2G 和 3G 升级到 LTE 来重复利用目前 2G 和 3G 频段。

此外，正如之前所提到的，相比使用相同功率的 2.5GHz 信号，Verizon and AT&T 计划使用的 700MHz 频段传输范围更广，穿透建筑物的能力更好。专家指出，700MHz 频段需要的基站数量是覆盖同等范围 2.5GHz 频段所需基站数的四分之一。

也许你已注意到，4G 技术的行业参与者代表了每一个标准的历史变化。一般情况下，LTE 的最大拥护者是电信服务公司和手机制造商，WiMAX 则更关注其支持者如 Intel, Cisco 和 Google 的喜好。值得注意的是，诸如 Nokia 和 Motorola 公司则同时是上述两个产业团体的拥护者，但其参与度有所不同。

Unit 11 Circuit-Switched Network and Packet-Switched Networks

11.1 Text

11.1.1 Circuit-switched networks

In ***circuit-switched networks***, such as the ***Public Switched Telephone Network*** (***PSTN***), multiple calls are transmitted across the same transmission medium. Frequently, the medium that is used in the PSTN is copper. However, fiber optic cable might also be used.

A circuit-switched network is a network in which there exists a dedicated connection. A dedicated connection is a circuit or channel that is set up between two nodes so that they can communicate. After a call is established between two nodes, the connection may be used only by these two nodes. When the call is ended by one of the nodes, the connection is canceled.

There are two basic types of circuit-switched networks: analog and digital. Analog was designed for voice transmission. For many years, the PSTN was only analog, but today, circuit-based networks such as the PSTN have transitioned from analog to digital. To support an analog voice transmission signal over a digital network, the analog transmission signal must be encoded or converted into a digital format before it enters the telephony WAN. On the receiving end of the connection, the digital signal must be decoded or converted back into an analog signal format.

There are advantages and disadvantages to circuit-switched networks. Circuit-switched networks have several disadvantages. Circuit-switched networks can be relatively inefficient, because bandwidth can be wasted. This is not the case when VoIP is used on a packet-switched network. VoIP shares the available bandwidth with all other network applications and makes more efficient use of the available bandwidth.

Circuit switching has one big advantage over packet-switched networks. In a circuit-switched network when you use a circuit, you have the full circuit for the time that you are using the circuit without competition from other users. This is not the case with packet switched networks.

11.1.2 Packet-Switched Networks

Packet switching is a technique that divides a data message into smaller units that are called packets. Packets are sent to their destination by the best route available, and then they are reassembled at the receiving end.

In ***packet-switched networks*** such as the Internet, packets are routed to their destination through the most expedient route, but not all packets traveling between two hosts travel the same route, even those from a single message. This almost guarantees that the packets will arrive at different times and out of order. In a packet-switched network, packets (messages or fragments of messages) are individually routed between nodes over data links that may be shared by other nodes. With packet switching, unlike circuit switching, multiple connections to nodes on the network share

the available bandwidth.

Packet-switched networks exist to enable data communication on the Internet throughout the world. A public data network or packet-switched network is the data counterpart to the PSTN.

Packet-switched networks are also found in such network environments as LAN and WAN networks. A WAN packet-switched environment relies on telephone circuits, but the circuits are arranged so that they retain a permanent connection with their end point. In a LAN packet-switched environment, such as with an ***Ethernet network***, the transmission of the data packets relies on packet switches, routers, and LAN cables. In a LAN, the switch establishes a connection between two segments only long enough to send the current packet. Incoming packets are saved to a temporary memory area or buffer in memory. In an ***Ethernet-based LAN***, an ***Ethernet frame*** contains the ***payload*** or data portion of the packet and a special header that includes the ***media access control* (*MAC*)** address information for the source and destination of the packet. When the packets arrive at their destination, they are put back in order by a packet assembler. A packet assembler is needed because of the different routes that the packets may take.

Packet-switched networking has made it possible for the Internet to exist and, at the same time, has made data networks—especially ***LAN-based IP networks***—more available and widespread.

11.1.3 Details of Packet Switching

Packet switching, described as Figure 11-1, is similar to message switching using short messages. Any message exceeding a network-defined maximum length is broken up into shorter units, known as packets, for transmission; the packets, each with an associated header, are then transmitted individually through the network. The fundamental difference in packet communication is that the data is formed into packets with a pre-defined header format (i.e. PCI), and well-known "idle" patterns which are used to occupy the link when there is no data to be communicated.

There are two important benefits from packet switching.

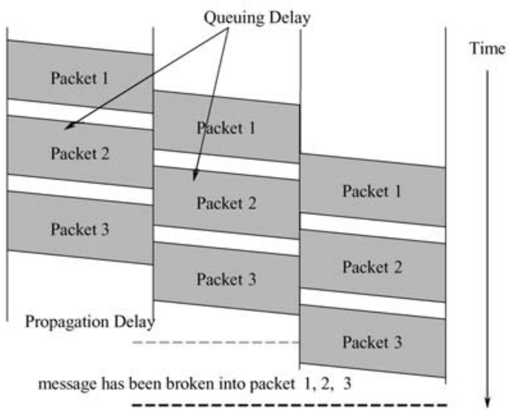


Figure 11-1 Packet-switched communication

1. The first and most important benefit is that since packets are short, the communication links between the nodes are only allocated to transferring a single message for a short period of time while transmitting each packet. Longer messages require a series of packets to be sent, but do not

require the link to be dedicated between the transmission of each packet. This provides a much fairer sharing of the resources of each of the links.

2. Another benefit of packet switching is known as “pipelining”. This simultaneous use of communications links represents a gain in efficiency, the total delay for transmission across a packet network may be considerably less than for message switching, despite the inclusion of a header in each packet rather than in each message.

11.1.4 Packet Switching & Circuit Switching

Compared with packet switching, circuit switching has advantages as well as disadvantages. Circuit switching employs ***physical connecting*** mode for relaying. Once the connection route has been set up, a special ***end-to-end communication path*** between the two terminals is assigned to the two users until the connection is canceled. Once the connection has been set up, it does not require any control signaling. Consequently, circuit switching embodies such essential features as following.

① Low transmitting delay;

② ***Transparent path*** to its subscribers;

③ Long setup time and low path utilizing ratio.

By allowing more than one user sharing a common transmitting path, the utilization ratio of the path in package switching is improved greatly. Because of the additional control information in each package data, there must be some ***propagation delay*** in package switching system.

Figure 11-2 shows the differences between a group of data transmitted by circuit and package switching separately. Supposing the transmitted message is a text with 40 8bit-code-groups, each package is assigned five 8bit-code-groups for control, which is shown as the black dot (Header) in Figure 11-2. The origin and destination of the message is A, B separately, while node1 and node2 are the relaying nodes along the transmission path.

In Figure11-2 (a), a single package which comprises 40 8bit-data is sent from A to B. Because of the additional header, this package has 45 8bit-data. Only when node1 receives all of these 45 bytes, can node1 relay them to node2. Similarly, only when node2 receives all of these data, can node2 relay them to B. $T_{\mathrm{A}\to 1}$、$T_{1\to 2}$、$T_{2\to \mathrm{B}}$ represent the “time from node A to 1”, “time from node 1 to 2”, and “time from node 2 to B” respectively, the total time of the transmission, namely $T_{(\mathrm{a})}$, according to Equation(11-1), is 135 8bit-data-transmitting time regardless of the switching time.

$$T_{(\mathrm{a})} = T_{\mathrm{A}\to 1} + T_{1\to 2} + T_{2\to \mathrm{B}} = (40+5)\times 3 = 135 \tag{11-1}$$

In Figure11-2 (b), the message sent to B is broken into two packages. Assuming the additional control header is still 5bits (actually a little more than 5bit) for each package, so each of two packages has 25 (40÷2＋5=25) 8bit-data. Because each package has its own control information, node1 may relay received data package to node2 immediately regardless of the arrival of the other package. Thus, the total transmitting time of the message from A to B, namely $T_{(\mathrm{a})}$, according to Equation (11-2), is 100 8bit-data-transmitting time.

$$T_{(\mathrm{b})} = (20+5)\times 4 = 100 \tag{11-2}$$

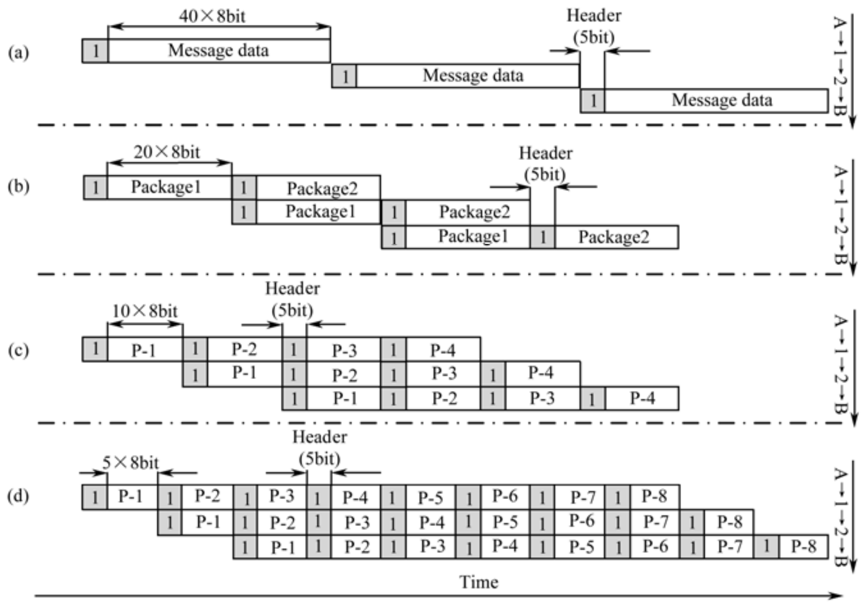


Figure 11-2 Circuit switching and package switching transmission

In Figure 11-2 (c), (d), the message sent to B is broken into 4, 8 packages respectively and the additional header is still 5bits for each. Each package is 15, 10 8bit-data respectively, and the corresponding transmitting time from A to B, namely $T_{(a)}$, according to Equation(11-3), (11-4), is 90, 100 8bit-data-transmitting time.

$$T_{(c)} = (10+5)\times 6 = 90 \tag{11-3}$$

$$T_{(d)} = (5+5)\times 10 = 100 \tag{11-4}$$

Among the above instances, the transmitting time of (c) is the shortest while time of (a) is the longest, from which we can draw the conclusion:

① Package transmission mode does not need to take account of the arrival of other packages at ***relaying nodes***, so the total message transmitting time is reduced by parallel transmission;

② Because of the ***additional POH*** for control information, for the same transmitted message, more packages do not mean less transmitting time, such as instance (d) in figure1.

Technical words and phrases

Public Switched Telephone Network (PSTN) 公用电话交换网

circuit-switched networks 电路交换网

copper ['kɔpə] *n.* 铜；铜币；铜制品

encode [in'kəud] *vt.* 编码

receiving end 接受端

provision [prə'viʒən] *vt.* 供给

route [ru:t] *n.* 路由

expedient [iks'pi:diənt] *adj.* 权宜的，方便的，有用的

out of order *adv.* 次序颠倒，不整齐，无序

fragment ['frægmənt] *n.* 数据包片段
router [ru:tə] *n.* 路由器
switch [switʃ] *n.* 交换机
buffer ['bʌfə] *n.* 缓冲区
relaying node 转接点
system-level 系统级
reassemble [ri:ə'sembl] *vt.* 再集合，再聚集；调整
packet switching 分组交换
packet-switched network 分组交换网
counterpart ['kauntəpɑ:t] *n.* 配对物，相应物
payload ['peiləud] *n.* 净载重量，有效负荷
additional POH 额外通道开销
Ethernet network 以太网
Ethernet-based LAN 以太局域网
LAN-based IP network 局域 IP 网
destination [,desti'neiʃən] *n.* 目的地
media access control (MAC) 媒体接入控制
control signaling 控制信令
control header 控制信头
Ethernet fram 以太网帧
physical connecting 物理连接
transparent path 路径透明
propagation delay 传播延迟

11.2 Reading Materials

1. Mobile Internet and GPS Change the Future of Smartphones in China

The year 2007 was a milestone for the smartphone, with a series of events impacting the future of the device. These events include the introduction of the Apple iPhone, Nokia's announcement of its Ovi Internet service portal, and Google's announcement of its Google Android Linux OS platform.

In-Stat believes that there are three key factors that will have the greatest impact on the Smartphone market.

① Mobile Internet. Riding a wave that is bringing mobile Internet to handsets, the smartphone is evolving to become a mobile Internet device, with cellular voice communication just one function of the converged appliance.

② Revolutionary UI and UE. Touchscreen UI and 3D sensor technologies are being integrated into smartphone operating systems to enable a more creative user experience.

③ Built-in GPS function. The GPS function enables the smartphone to be a mobile navigation

device and provide location-based service to mobile users.

2. Chinese ULC Phone Market Will Peak in 2008

Driven by declines in handset prices and voice communication fees, there will be over 80 million new wireless subscribers each year in China between 2008 and 2012. In-Stat believes that this indicates a huge potential market for ULC phones.

Nokia's and Motorola's success in the ULC handset sector is driving other competitors to enter the market. ULC phones are targeted at three groups of consumers: low-income people in rural or urban areas (for example, migrant workers) and young students who are financially supported by their families; mobile users who only need the call function and are unable to or uninterested in using the extended functions of mobile phones (this group includes the elderly and children); and PHS phone users who want better communication quality or roaming services.

3. China Mobile Serves 2008 Olympics

As the 2008 Beijing Olympics drawing nearer, the Chinese government is testing wireless local area networks (WLAN) to determine which one is safest.

China Mobile, the biggest mobile telecom carrier in China and the mobile communication partner of the Beijing 2008 Olympics, is carrying out the tests.

Test results to date show that the current WLAN technology 802.11i has big security loopholes and is easy to attack, said Ma Benteng, senior engineer with China Mobile.

The Beijing Olympics will be the first to use WLAN in the Games' history. Journalists would be major users of the networks.

At a meeting held by China Mobile recently, media users were skeptical about the safety of the current WLAN technology.

Results from more than a month of tests carried out by the national safety research center on information project show that 802.11i has serious technological defects and safety risks, said Ma, who is in charge of mobile planning for the 2008 Olympics.

Researchers said that articles on the technological defects of 802.11i were freely available on the internet, as well as tools for exploiting the defects. The internet also provides methods for decoding the technology.

Anybody who can connect to the Internet could download the software and steal private information from others, said Ma.

That would potentially cause users huge losses, especially media users whose Olympics stories, photos or visual clips could be stolen during transmission, said Ma.

As system operator, China Mobile would be expected to assume some of the responsibility if this were to happen. Ma called for concerned departments and companies to pay close attention to the safety issue and propose safer technologies.

The current WLAN technology, based on the 802.11 series, has drawn criticism from experts and customers because of its safety loopholes. At a seminar held on Sept. 11 this year, researchers used tools downloaded from the Internet and decoded protection passwords in just five minutes.

Intel and IWNCOMM, a private Chinese company, have separately developed 802.11i and

WAPI to remedy safety defects.

Analysts said that 802.11i's poor test performance may give China's WAPI an opportunity. WAPI was adopted as China's national standard in 2003.

China Mobile was officially established on April 20, 2000 and is directly under the administration of the central government. It is a key state-owned telecom comapny based on the mobile business that split from the former China Telecom as a result of the reform and restructuring of China's communications industry.

China Mobile Communications Corporation has a registered capital of CNY 51.8 billion, assets of over CNY 320 billion and 120,800 employees. It has wholly-owned subsidiaries in 31 provinces (autonomous regions) in China and fully holds the equity of China Mobile (HK) Group Limited.

(source:SINOCAST)

4. ITU to hold Next Generation Network Workshop

Geneva, 21 April 2005 — ITU will hold a workshop on next generation networks (NGN) together with the Internet Engineering Task Force (IETF), 1-2 May, 2005, Geneva.

Since May 2004 intense work has taken place in ITU, towards the development of standards that will define services, network and systems architecture in the next generation of IP enabled communication systems, or next generation networks (NGN).

The objectives of the workshop are to report the progress of ITU’s work on NGN and explore specific issues that impact both the ITU and the IETF in order to better understand the work underway in the two organizations and to identify areas where action can be taken to make further progress.

Houlin Zhao, Director of the ITU’s Telecommunication Standardization Bureau notes that, “We have made tremendous progress, thanks to the support of ITU members and members of other standards developing organizations such as IETF, ETSI[1] and ATIS[2]. The momentum that this work has achieved will allow the ICT industry to develop a raft of new products and services on a much more powerful and dynamic infrastructure based on globally accepted standards.”

Six sessions will each be co-chaired by an ITU representative and a representative from IETF. One area to be addressed is the concept known as “nomadicity”, which will give fixed line and mobile users completely seamless communication. Simply put this means the underlying technology will be invisible to the user in a multi-service, multi-protocol, multi-vendor environment. The increasingly important topic of security will also be examined, along with signaling, QoS, requirements and functional architecture, network management and the evolution from the traditional circuit switched network.

5. Mobile IM in China: Call for Enabled Handsets and Applicable Client Software

Instant Messaging (IM) service has grown with the Internet. From 2000 to 2005, the Short Messaging Service (SMS) model of IM service was considered an extension of IM for the PC. In 2005, however, wireless carriers China Mobile and China Unicom began to pay greater attention to IM service. They each introduced a trial version of their IM client software in the second half of

2006, and launched full mobile IM client applications in mid-2007. To compete with carriers, IM services providers such as Tencent and Microsoft have also released IM client products. The IM market is once again fiercely competitive, but this time for handsets rather than PCs.

To help these industry players learn more about the attitudes of customers toward IM, In-Stat carried out a web-based survey in March 2007 aimed at studying levels of interest and patterns of use among consumers.

The survey demonstrates that the mobile IM market has entered a mature, mass-market stage, since a high proportion of respondents who were either positive or neutral to new technology had used the service. We also found that non-users had a moderate level of interest in mobile IM: 19.8% of respondents who had not used it said they were either extremely or very interested in the service.

This report surveyed 725 respondents, of whom 374, or 51.6%, had not previously used mobile IM, and of whom 351, or 48.4%, were current users.

11.3 Exercises

1. Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）电路交换网
（2）路由器
（3）缓冲
（4）媒体接入控制
（5）转接点
（6）公用电话交换网
（7）系统级
（8）分组交换网
（9）额外通道开销
（10）传播延迟
（11）以太局域网
（12）局域 IP 网
（13）以太网
（14）以太网帧
（15）物理连接
（16）路径透明

2. Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）In circuit-switched networks multiple calls are transmitted across the same(　　　) medium.
（2）A circuit-switched network is a network in which there exists a (　　　) connection.
（3）There are two basic types of circuit-switched networks(　　　) and (　　　　).
（4）Packet switching is a technique that divides a data message into smaller units that are called

(　　).

（5）Packet-switched networks are also found in such network environments as (　　) and (　　) networks. A WAN packet-switched environment relies on (　　) circuits. In a LAN packet-switched environment, the transmission of the data packets relies on (　　) switches, routers, and LAN cables.

（6）The cellular concept was a major breakthrough in solving the problem of（　　）and (　　).

（7）As the demand for service increases in cellular communication system, the number of base stations may be (　　), thereby providing additional radio (　　) with no additional increase in radio (　　).

（8）Cellular radio systems rely on an intelligent (　　) and (　　) of channels throughout a coverage region.

（9）The design process of selecting and allocating channel groups for all of the cellular base stations within a system is called (　　) or (　　).

（10）The mobile station contains a (　　), an (　　), and (　　).

（11）Communication between the base station and the mobiles is defined by (　　) that specifies four different channels.

（12）In all cellular systems, a kind of service which is called (　　) allows subscribers to operate in service areas other than the one from which service is subscribed.

3. Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）(　　) Frequently, the medium that is used in the PSTN is copper.

（2）(　　)After a call is established between two nodes, the connection may be used only by these two nodes.

（3）(　　) Today, circuit-based networks such as the PSTN was still analog.

（4）(　　) To support an analog voice transmission signal over a digital network, the analog transmission signal must be encoded or converted into a digital format.

（5）(　　) Circuit-switched networks are more efficient than packet-switched network.

（6）(　　) In circuit switching system, once the connection has been set up, it does not require any control signaling.

（7）(　　) The utilization ratio of the path in package switching is not improved greatly.

（8）(　　) Total message transmitting time is reduced by serial transmission in package transmission.

（9）(　　) The cellular concept calls for replacing many low power transmitters (small cells) with a single high power transmitter (large cell).

（10）(　　) The transmitters and receivers in base stations are capable of handling full duplex communications.

11.4　课文参考译文　电路交换和分组交换

11.4.1　电路交换

电路交换网络，如公共电话交换网（PSTN）中，多路呼叫（信号）同时在同一条线路上传输。通常情况下，PSTN 网络所使用的线路传输介质是金属铜制成的，此外也可采用光纤作为传输媒介。

电路交换网络中，为用户呼叫而建立的链接是一条专用的电路通道，它建立在两个节点之间，在整个通话期间始终保持连接状态，从而保证了两点之间的通信。当两点间的呼叫通路建立后，该专用连接通道只限于这两个节点使用，若其中任意一个节点提出终止通信的请求，这个连接即被中断。

电路交换网络有两种基本类型：模拟交换网络和数字交换网络。模拟交换网络主要用来传输语音。许多年来，PSTN 网络只使用模拟交换方式，但是现在，像 PSTN 的电路交换网络已经从模拟网转换成数字网了。为了实现在数字化网络中传输模拟的语音信号，在信号进入电话网之前，模拟信号必须经过编码并被转换成数字形式。在呼叫到达被叫方，数字信号必须经过解码并被转换成模拟形式。

电路交换技术有其优缺点。电路交换的缺点是存在着带宽的浪费这一现象，所以其利用率不高。而对于 VoIP 网络而言，由于其利用了分组交换技术，所有的网络应用都可以共同来使用可用带宽，所以带宽利用率较高。

电路交换技术相比于分组交换也有一个优点。在电路交换网络里，在你使用电路通道的这段时间里，你完全占有了这条电路通道，而不会受到其他用户的影响。在分组交换网络里可不是这样的。

11.4.2　分组交换

分组交换技术将数据划分成更小的单元，称为数据包。数据包以最佳的路由被转发到目的地。在接收端这些数据包被重组。

在分组交换网里，如互联网，数据包以最优的路径被路由到目的地，但是，不是说所有的数据包都是沿着同样的路径传输的，即使这些数据包来自于同一条数据消息。这样的话数据包就不会在相同时间到达，同时在到达时刻它们也是无序的。在电路交换网里，各个数据包（消息或消息的一部分）在两个节点之间的数据链路上被分别转发，这条数据链路也可能被其他节点所使用。不同于电路交换，在分组交换网中，到某个节点的数个数据链路是可以共享使用的。

分组交换的存在使得通过互联网实现全球范围内的数据通信称为了可能。公众数据网络或分组交换网络对应于 PSTN 网络中的数据传输部分。

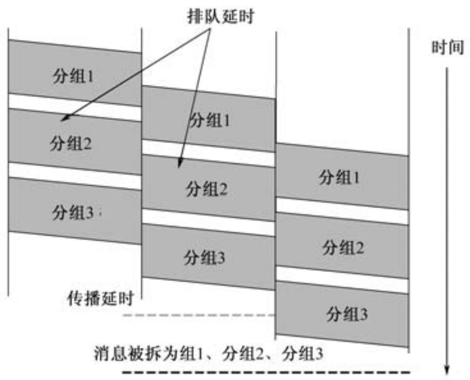
分组交换网络同样也在局域网和广域网中得到应用。广域网中分组交换依赖于电话的电路通道，但是该电路通道与终端是始终保持连接状态的。在基于局域网的分组交换环境中，如以太网，数据包的路由依赖于分组交换、路由和局域网中的电缆。在局域网中，在分段的两个节点之间建立链接，分段的长度刚好可以传输当前的数据包。到达的数据包保存在临时的存储器里或缓冲里。在基于以太网的局域网里，一个以太网的数据帧包含数据包的载荷和数据部分，同时该数据帧含有一个特殊的报头，包含了该数据包源和目的地的媒体接入控制（MAC）地址的信息。当数据包到达目的地时，它们通过数据包排序被重组。数据包排序

是必须的，因为不同的数据包会沿不同的路径到达。

分组交换网络使得互联网的出现成为了可能。与此同时，使数据网络——尤其是基于局域网的 IP 网络——的普及成为了可能。

11.4.3 分组交换的一些细节

图 11-1 所示的分组交换使用了短消息模式。任何超过网络规定最大长度的消息被分解成更小的单元，称之为数据包，以便传输。每一个数据包都有一个相应的报头，数个数据包各自通过网络传输出去。分组交换最重要的一个特点就是，数据消息被转换为数据包，每一个数据包携带一个预先定义好的报头格式（就是 PCI），如果没有数据传输，采用我们熟知的“闲置”模式来占有通信链路。



译图 11-1 分组交换的通信方式

分组交换有两个优点：

最显著的优点就是因为数据包很短，在两个节点之间的通信链路只会给每个数据包分配很短的时间进行传输。更长的数据消息需要一长串的数据包实现传输。在数据包传输时，该通信链路并没有被独占。这样通信链路的共享方式就显得更加公平

分组交换的另一个显著的特点就是熟知的“流水操作”。来自于不同呼叫的各个数据包同时使用通信链路，使得效率得到了提高。尽管在每一个数据包里均添加了报头，分组交换网络中的传输总延时比消息交换的时间要小得多。

11.4.4 分组交换与电路交换性能对比

分组交换和电路交换相比有优点也有缺陷。电路交换的接续路径采用物理连接方式，一旦接通就形成一条端对端（用户终端和被叫终端之间）的专用通路，直到线路拆除前仅供这一对终端使用，也不再需要控制信息。所以，电路交换具有如下基本特点：

① 信息传输延迟小；

② 向用户提供的通路是透明的；

③ 电路接续时间长、线路利用率低。

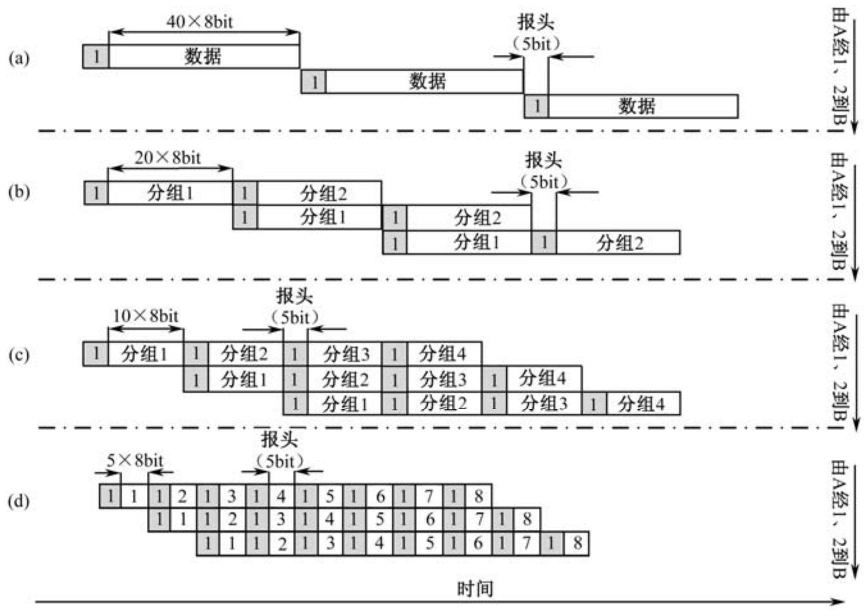
分组交换方式使多个用户共享一条传输线路，线路利用率必然大大提高；但在分组数据中增加了控制信息，会产生一定的分发延迟。

图 11-2 所示是一组数据分别采用电路交换和分组交换进行传输的情况对比。设发送的报文为 40 个 8bit 码组、分组传送时每个分组均包含 5 个 8 位控制信息，即图中黑色小框表示的报头（Header）。由 A 端发送到目的地 B 端，首先要经过站点 1，由站点 1 发送到站点 2，最后才可由 2 到达目的 B。

图 11-2（a）所示为 40 个 8 位数据作为一个分组整体发送到终点的情况。由于加上了报头，该分组具有 45 个 8 位数据，只有站点 1 完全收到这 45 个 8 位组后才能向站点 2 发送；同样，站点 2 也必须完全接收到该 45×8 位数据后才向 B 发送。不考虑其他交换时间，用

$T_{A\to1}$、$T_{1\to2}$、$T_{2\to B}$分别表示“端点 A 到站点 1 的时间”、“站点 1 到站点 2 的时间”、“站点 2 到端点 B 的时间”，则完成这一过程所需总时间$T_{(a)}$由式（11-1）求得，为 135 个 8bit 的数据传输时间。

$$T_{(a)} = T_{A\to1} + T_{1\to2} + T_{2\to B} = (40+5)\times3 = 135 \tag{11-1}$$



译图 11-2　电路交换和分组交换传输分析

图 11-2（b）所示为将报文分两个分组发送的情况。设每个分组所需控制信息仍是 5 位（实际中可能会略多于此），则两个分组大小就是各 40÷2＋5＝25 个 8 位。由于每个分组均包含控制信息，站点 1 可以在收到一个分组后立即按其包头信息发往站点 2，而不必等待收齐分组后再一起转发。这样，传输过程中出现了并行情况，由式（11-2）求得总传输时间$T_{(b)}$为 100 个 8 位时间。

$$T_{(b)} = (20+5)\times4 = 100 \tag{11-2}$$

与此类似，图 11-2（c）、（d）分别表示将报文分成 4 个、8 个分组传送的情况。此时每个分组的大小分别为 15、10 个 8 位，相应的总传输时间$T_{(c)}$、$T_{(d)}$分别由式（11-3）、式（11-4）求得为 90、100 个 8 位时间。

$$T_{(c)} = (10+5)\times6 = 90 \tag{11-3}$$

$$T_{(d)} = (5+5)\times10 = 100 \tag{11-4}$$

上述 4 种情况下，图（c）的传输时间最短，而图（a）最长，说明：

① 分组传输不必在中间站点等待分组到齐，从而使传输时间因并行而得以缩短；

② 并非分组越多，总传输时间就越短，这是因为分组需添加控制信息而产生额外开销，如上述图（d）所示情况。

Unit 12 An Overview of Fiber Optic Technology

12.1 Text

The use of ***fiber optics*** in telecommunications and wide area networking has been common for many years, but more recently fiber optics have become increasingly prevalent in industrial data communications systems as well. High data rate capabilities, noise rejection and electrical isolation are just a few of the important characteristics that make fiber optic technology ideal for use in industrial and commercial systems.

Most often used for point-to-point connections, fiber optic links are being used to extend the distance limitations of RS-232, RS-422/485 and ***Ethernet*** systems while ensuring high data rates and minimizing electrical interference. Conventional electrical data signals are converted into a modulated light beam, introduced into the fiber and transported via a very small diameter glass or plastic fiber to a receiver that converts the light back into electrical signals. Fiber's ability to carry the light signal, with very low losses, is based on some fundamental physics associated with the refraction and reflection of light.

12.1.1 Fiber Optic Principles

Whenever a ray of light passes from one transparent medium to another, the light is affected by the interface between the two materials. This occurs because of the difference in speeds that the light can travel through different materials. Each material can be described in terms of its ***refractive index***, which is the ratio of the speed of light in the material to its speed in free space. The relationship between these two refractive indices determines the critical angle of the interface between the two materials.

There are three actions that can happen when a ray of light hits an ***interface***. Each action depends on the angle of incidence of the ray of light with the interface. If the angle of incidence is less than the critical angle, the light ray will refract, bending toward the material with the higher refractive index. If the angle of incidence is exactly equal to the critical angle the ray of light will travel along the surface of the interface. If the angle of incidence is greater than the critical angle, the ray of light will reflect.

The refractive index of vacuum is considered to be 1. Often, we consider the refractive index of air also to be 1(although it is actually slightly higher). The refractive index of water is typically about 1.33. Glass has a refractive index in the range of 1.5, a value that can be manipulated by controlling the composition of the glass itself.

12.1.2 Fiber Optic Characteristics

Optical fibers allow data signals to propagate through them by ensuring that the light signal enters the fiber at an angle greater than the critical angle of the interface between two types of glass. As shown in Figure 1, optical fiber is actually made up of three parts. The center ***core*** is composed

of very pure glass, with a refractive index of 1.5. Core dimensions are usually in the range of 50 to 125 um. The surrounding glass, called ***cladding***, is a slightly less pure glass with a refractive index of 1.45. The diameter of the core and cladding together is in the range of 125 to 440 um. Surrounding the cladding is a protective layer of flexible silicone called the ***sheath***.

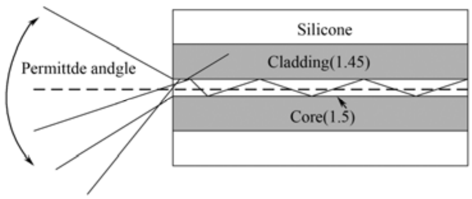


Figure 12-1 Light Traveling Through a Fiber

When light is introduced into the end of an optical fiber, any ray of light that hits the end of the fiber at an angle greater than the critical angle will propagate through the fiber. Each time it hits the interface between the core and the cladding it is reflected back into the fiber. The angle of acceptance for the fiber is determined by the critical angle of the interface. If this angle is rotated, a cone is generated. Any light falling on the end of the fiber within this cone of acceptance will travel through the fiber. Once the light is inside the fiber it 'bounces' through the core, reflecting inward each time it hits the interface.

Figure 12-1 illustrates how light rays travel through the fiber, reflecting off the interface. If the physical dimensions of the core are relatively large individual rays of light will enter at slightly different angles and will reflect at different angles. Since they travel different paths through the fiber, the distance they travel also varies. As a result they arrive at the receiver at different times. A ***pulse signal*** sent through the fiber will emerge wider than it was sent, deteriorating the quality of the signal. This is called ***modal dispersion***. Another effect that causes deterioration of the signal is ***chromatic dispersion***. Chromatic dispersion is caused by light rays of different wavelengths traveling at different speeds through the fiber. When a series of pulses is sent through the fiber, modal and chromatic dispersion can eventually cause the pulse to merge into one long pulse and the data signal is lost.

Another characteristic of optical fiber is ***attenuation***. Although the glass used in the core of optical fiber is extremely pure, it is not perfect. As a result light can be absorbed within the cable. Other signal losses include bending and scattering losses as well as losses due to connections. Connection loses can be caused by misalignment of the ends of the fiber or end surfaces not properly polished.

12.1.3 Types of Fibers

Optical fibers are manufactured in three main types: ***multi-mode step-index***, ***multi-mode graded-index***, and ***singlemode***. Multi-mode step-index fiber has the largest diameter core (typically 50 to 100 um). The larger distance between interfaces allows the light rays to travel the most distance when bouncing through the cable. Multi-mode fibers typically carry signals with wavelengths of 850 nm or 1300 nm. Figure 12-2 shows how a narrow pulse introduced to the fiber

becomes wider at the receiving end.

Multi-mode step-index fiber, as shown as Figure 12-2 (a), is comparatively easy to splice and terminate due to the large diameter fiber. It is also relatively inexpensive to manufacture compared to other types. However, it tends to be too slow for most purposes and it is not common in modern systems.

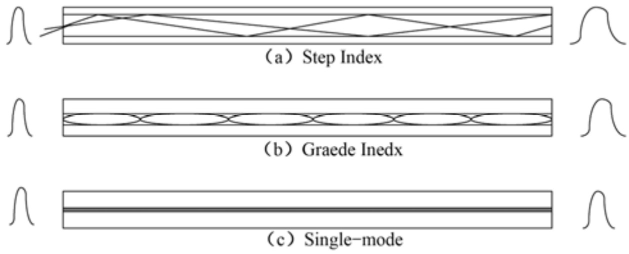


Figure 12-2　A narrow pulse introduced to various fibers

Multi-mode graded-index fiber, as shown as Figure 12-2 (b), is constructed in such a way that the refractive index between the core and cladding changes gradually. This causes the light rays to bend gradually, as well. The resulting pattern of reflections tends to be more uniform and dispersion is reduced. This provides improved performance for a moderate increase in cost. Graded-index fibers provide wider bandwidth than step-index fibers.

Single-mode fibers, as shown as Figure 12-2 (c), give the highest performance of the three types. Manufactured using a very small diameter fiber (typically 8 um), when light is introduced into the fiber reflections are kept to a minimum by the dimensions of the core. Light travels virtually straight through the core and pulses introduced at one end are reproduced at the other end with very little dispersion. Typically, single-mode fibers carry signals with wavelengths of 1320 nm or 1550 nm. Single-mode fiber is relatively expensive, however, and is more difficult to splice and terminate since the core must be aligned very accurately.

Single-mode fibers offer much lower attenuation than multi-mode fibers. At typical single-mode fiber will attenuate a 1310 nm signal less than 0.5 dB per kilometer. A typical multi-mode graded-index fiber will attenuate the same signal about 3 dB per kilometer. Single-mode fiber is most often used in applications with high bandwidth requirements over long distances. Some Ethernet fiber optic equipment can increase distances from two kilometers using multi-mode fiber to about 70 kilometers over single-mode fiber.

12.1.4　Advantages of Fiber Optic Cables

Noise Immunity

Noise immunity is one of the most useful features of fiber optics in industrial applications. In environments where electromagnetic interference is prominent and unavoidable, fiber optics are unaffected. While cables are normally contained in protective sheaths and often run inside conduit, there is no need to physically isolate fiber optic cables from electrical cables. This makes cable routing simpler.

Electrical Isolation

The problem of ground loop noise and common mode potential differences is eliminated by the use of fiber optic cables. Field signals, generated by devices floating at high potentials, can be coupled to other equipment at much lower potentials without the risk of damage. This is particularly desirable in industrial applications.

Low Error Rates

When properly designed to provide adequate signal levels at the receiving end of the link, a fiber optic system provides very low bit error rates.

Safe for Use in Hazardous Areas

Fiber optic links can be used to couple signals into areas with potentially explosive atmospheres without a risk to delivering or storing sufficient energy to ignite an explosion. This makes fiber optic technology particularly useful when designing intrinsically safe systems.

Wide Bandwidth

Fiber optic cables can carry very wide bandwidth signals, well into the GHz range. Many individual, lower bandwidth signals can be multiplexed onto the same cable. In commercial systems fiber optic cable often carries a mixture of signal types, including voice, video and data all on the same fiber.

Low Signal Attenuation

Optical fibers do exhibit some attenuation due to absorption and scattering. However, this attenuation is relatively independent of frequency, a factor that is significant in copper cables.

Light Weight, Small Diameter

Because many signals can be multiplexed onto one fiber, cables tend to be smaller and lighter. This makes installation easier.

No Crosstalk

Since fibers do not pick up electromagnetic interference, signals on adjacent cables are not coupled together.

Inherent Signal Security

For applications where signal security is a concern, optical fiber is an excellent solution. Fiber optic cables do not generate electromagnetic fields that could be picked up by external sensors. It is also more difficult to 'steal' signals by spicing into optical fibers than it might be with conventional copper wiring.

Technical words and phrases

optics ['ɔptiks] *n.* 光学
fiber optics *n.* 光纤
link [liŋk] *n.* 链路
Ethernet 以太网
prevalent ['prevələnt] *adj.* 流行的
interface ['intə(:)feis] *n.* 界面，接触面
noise rejection 噪声抑制，去噪，噪声剔除
isolation [aisə'leiʃən] *n.* 隔绝，孤立，隔离
electrical isolation 电隔离，电气隔离
interference [intə'fiərəns] *n.* 干涉，干扰
electrical interference 电磁干扰
light beam 光束
refractive index 折射率
angle ['æŋgl] *n.* 角，角落
critical angle 临界角
angle of incidence 入射角
loss [lɔ(:)s] *n.* 损失
material [mə'tiəriəl] *n.* 材料
pure [pjuə] *a.* 纯净的，完美的，无瑕的
dimension [di'menʃən] *n.* 尺寸
pulse [pʌls] *n.* 脉冲
pulse signal 脉冲信号
cladding ['klædiŋ] *n.* 包层
core [kɔ:] *n.* 纤芯
sheath [ʃi:θ] *n.* 护套
cone [kəun] *n.* 锥，圆锥
modal dispersion 模间色散，模式色散
chromatic dispersion 色散，色度色散
attenuation [ətenju'eiʃən] *n.* 衰减
bending and scattering losses 弯曲和散射损耗
misalignment ['misəlainmənt] *n.* 不重合
manufacture [mænju'fæktʃə] *vt.* 制造,加工
step-index 阶跃折射率
graded-index 梯度折射率
multi-mode step-index fiber 多模阶跃光纤
multi-mode graded-index fiber 多模梯度光纤
singlemode 单模

12.2 Reading Materials

1. The Optical Fiber-Losses in Optical Fiber

Other than the losses exhibited when coupling LEDs or LDs into a fiber, there are losses that occur as the light travels through the actual fiber.

The core of an optical fiber is made of ultra-pure low-loss glass. Considering that light has to pass through thousands of feet or more of fiber core, the purity of the glass must be extremely high. To appreciate the purity of this glass, consider the glass in common windowpanes. We think of windowpanes as “clear,” allowing light to pass freely through, but this is because they are only 1/16 to 1/4 inch thick. In contrast to this clear appearance, the edges of a broken windowpane look green and almost opaque. In this case light is passing edgewise into the glass, through several inches. Just imagine how little light would be able to pass through a thousand feet of window glass!

Most general purpose optical fiber exhibits losses of 4 to 6 dB per km (a 60% to 75% loss per km) at a wavelength of 850nm. When the wavelength is changed to 1300nm, the loss drops to about 3 to 4 dB (50% to 60%) per km. At 1550nm, it is even lower. Premium fibers are available with loss figures of 3 dB (50%) per km at 850nm and 1 dB (20%) per km at 1300nm. Losses of 0.5 dB (10%) per km at 1550 nm are not uncommon. These losses are primarily the result of random scattering of light and absorption by actual impurities within the glass.

Another source of loss within the fiber is due to excessive bending, which causes some of the light to leave the core area of the fiber. The smaller the bend radius, the greater the loss. Because of this, bends along a fiber optic cable should have a turning radius of at least an inch.

2. The Optical Fiber: Optical Fiber Bandwidth

All of the above attenuation factors result in simple attenuation that is independent of bandwidth. In other words, a 3 dB loss means that 50% of the light will be lost whether it is being modulated at 10 Hz or 100 MHz.

There is an actual bandwidth limitation of optical fiber however, and this is measured in MHz per km. The easiest way to understand why this loss occurs is to refer to Figure 12-3.

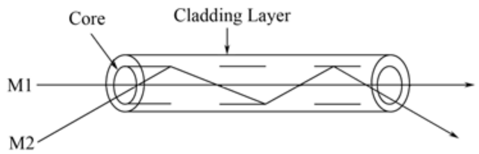


Figure 12-3 Different Light Path Lengths Determine the Bandwidth of a Fiber

3. The AOE（Asia Optical Fiber Communication& Optoelectronic Exposition）Conference

Starting in 2007, the AOE technical conference is now co-sponsored by the IEEE Lasers and Electro-Optics Society (IEEE LEOS) and the Optical Society of America (OSA). The expanded technical focus of the conference includes (1) Optical Fibers, Fiber Components and Subsystems, (2) Optoelectronic Devices and Materials, (3) Optical Sensors and Biophotonics, and (4) Displays,

Solid-State Lighting, and Optoelectronics in Energy.

The AOE Conference provides an ideal venue to catch up with new research directions, the latest technical breakthroughs, and emerging new commercial applications of optoelectronics subsystems and technologies. The conference features a full suite of plenary talks, invited talks, contributed talks and posters given by today's top international academic and industrial research leaders in their respective fields.

4. Wireless Local Loop (WLL)

The rapid growth of the Internet has created a concurrent demand for broadband Internet and computer access from businesses homes throughout the world. Particularly in developing nations where there is inadequate telecommunications backbone infrastructure, there is a tremendous need for inexpensive, reliable, rapidly deployable broadband connectivity that can bring individuals and enterprises into the information age. In fact, as voice over Internet protocols (VoIP) become prevalent, it is quite conceivable that a single broadband Internet connection could someday provide all of the needed telecommunications services, including telephone service, television, radio, fax, and Internet, for a home or business customer.

Fixed wireless communication systems are able to take advantage of the very well-defined, time-invariant nature of the propagation channel between the fixed transmitter and fixed receiver. Furthermore, modern fixed wireless systems are usually assigned microwave or millimeter radio frequencies in the 28GHz band and higher, which is greater than ten times the carrier frequency of 3G terrestrial cellular telephone networks. At higher frequencies, more bandwidth can be easily used.

Microwave wireless links can be used to create a wireless local loop (WLL) such as the one shown in Figure 12-4. In most developed countries, copper or fiber optic cable already has been installed to residences and business. However, in many developing nations, cable is too expensive to take or can take months or years to install. Wireless equipment, on the other hand, can usually be deployed in just a couple of hours. An additional benefit of WLL technology is that once the wireless equipment is paid for, there are no additional costs for transport for between the CO and the customer premises equipment (CPE), whereas buried cables often must be leased from a service provider or utility company on a monthly basis.

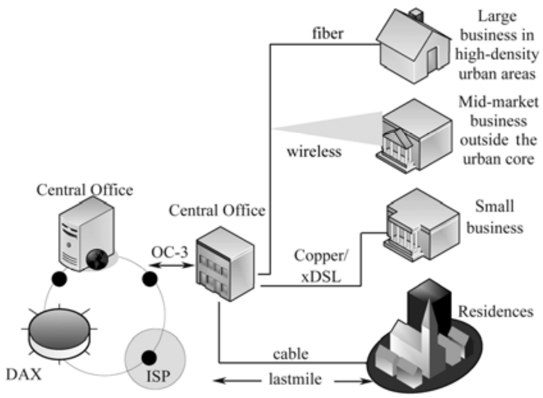


Figure 12-4 Emerging applications and markets for broadband services

5. Local Multipoint Distribution Service (LMDS)

Governments throughout the world have realized that WLL could greatly improve the efficiency of their citizens while stimulating competition that could lead to improved telecommunications services. A vast array of new services and applications have been proposed and are in the early stage of commercialization. These services include the concept of Local Multipoint Distribution Service (LMDS), which provides broadband telecommunications access in the local exchange.

The US LMDS band is 27.5～28.35 GHz, 29.1～29.25 GHz, and 31.075～31.225 GHz. The IEEE 802.16 Standards Committee is developing interoperability standard for fixed broadband wireless access. In Europe, a similar standard, HIPERACCESS, is being developed by a standardization committee for Broadband Radio Access Networks (BRAN) for operation in the 40.5～43.5GHz; it will use TDMA. Also, HIPERLINK is very high speed short range interconnection for HIPERLANS and HIPERACCESS, up to 155Mbit/s within 150 meters, and is planned to operate in the 17GHz band in Europe.

The US cellular service was first issued spectrum in 1983 and currently occupies 50MHz of the total bandwidth. The network in Figure 12-5 shows the current popular structure for wireless exchange. The PCS services occupy 150 MHz of bandwidth, and the unlicensed National Information Infrastructure (UNII) band, occupies 300MHz. LMDS, on the other hand, was allocated a whopping 1300MHz of bandwidth-enough spectrum to provide over 200 broadcast quality television channels or 65000 full duplex voice channels! Yet, the revenues generated from the auction of all the US LMDS licenses was a paltry ＄500 million, compared to the more than ＄30 billion generated from the PCS auctions held three years earlier! This market difference is due to the fact that LMDS is a brand new type of service, vastly unproven, and dependant upon millimeter wave equipment that is still expensive!

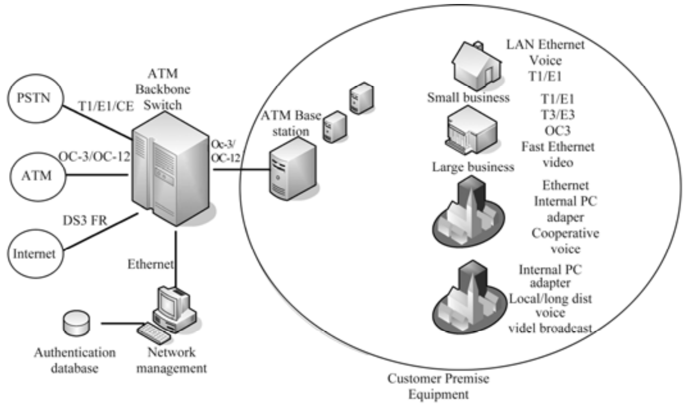


Figure 12-5　A wireless CLEC using ATM distribution

12.3 Exercises

1．Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）折射率
（2）入射角
（3）临界角
（4）纤芯
（5）包层
（6）模间色散
（7）单模光纤
（8）多模光纤
（9）本地无线环路
（10）本地多点分配业务

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）The characteristics that make fiber optic technology ideal for use in industrial and commercial systems are ( ), ( ) and ( ).

（2）Whenever a ray of light passes from one transparent medium to another, the light is affected by the ( ) between the two materials. This occurs because of the difference in ( ) that the light can travel through different materials.

（3）There are three actions that can happen when a ray of light hits an interface. If the angle of incidence is less than the critical angle, the light ray will ( ). If the angle of incidence is exactly equal to the critical angle the ray of light will travel along the ( ) of the interface. If the angle of incidence is greater than the critical angle, the ray of light will ( ).

（4）The refractive index of vacuum is considered to be ( ). The refractive index of air is considered be ( ). The refractive index of water is about ( ). Glass has a refractive index in the range of ( ).

（5）The angle of acceptance for the fiber is determined by the ( ) of the interface. If this angle is rotated, a ( ) is generated.

（6）Optical fibers are manufactured in three types: multi-mode （ ）, multi-mode （ ）, and （ ）.

（7）WLL represents the abbreviation of ( ).

（8）LMDS represents the abbreviation of ( ).

（9）The PCS services occupy ( ) MHz of bandwidth, the National Information Infrastructure band occupies ( ) MHz of bandwidth and LMDS occupies ( ) MHz of bandwidth.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）（　　）If the angle of incidence is greater than the critical angle, the ray of light will refract.
（2）（　　）The refractive index of air is consider to be 1，but it is actually slightly higher than 1.
（3）（　　）Glass has a refractive index in the range of 1.5, a value that can not be changed.
（4）（　　）Core dimensions of fiber optic are usually in the range of 50 to 125 um.
（5）（　　）The protective layer surrounding the cladding is called the core.
（6）（　　）Multi-mode step-index fiber is relatively inexpensive to manufacture.
（7）（　　）Graded-index fibers provide wider bandwidth than step-index fibers.
（8）（　　）Single-mode fiber is relatively inexpensive; however, it is more difficult to splice and terminate.
（9）（　　）Single-mode fibers offer much lower attenuation than multi-mode fibers.
（10）（　　）It is impossible that a single broadband Internet connection could provide all of the needed telecommunications services.
（11）（　　）Cable is too expensive to take or can take months or years to install
（12）（　　）The market difference between LMDS and PCS is due to the fact that LMDS is a brand new type of service, vastly unproven, and dependant upon expensive millimeter wave equipment.

12.4　课文参考译文　光纤技术概览

光纤在电信和广域网中投入使用有很多年了，但只是在近几年，光纤在数据通信系统中的使用才普及开来。高数据率、抗噪声和电气隔离特性是光纤之所以被工业和商业系统广泛使用的原因之一。

光纤更多地用于点对点的连接，在保证高数据率和电气干扰最低的同时，光纤突破了RS-232、RS-422/485 和以太网的传输距离限制。传统的电气数据信号被转换成一道调制的光束，导入到光纤，并在这根直径非常细的玻璃或塑料制成的光纤内传输到接收端，在接收端光纤被转换回电气信号。光纤的低损耗传输能力是基于光在折射和反射上的物理特性所带来的。

12.4.1　光纤的原理

当一束光从一种透明的介质传递到另一种透明介质时，光束将受到两种介质之间的界面影响。这是因为光在不同介质内部传输的速度不同所致。每一种介质都可以用折射率来表示，即光在该介质中的速度与光在自由空间中的速度之比。两种介质折射率的关系决定了这两种介质之间的临界角。

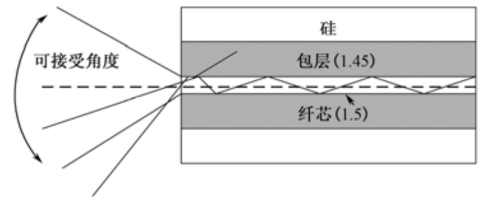
当一道光束入射到两种介质的界面时，有三种情况可能发生。每一种发生情况取决于光束与界面之间的夹角。如果入射角小于临界角，光束将沿着折射率更高介质折射进去。如果入射角正好等于临界角，光束将沿着这两种介质的表面传播下去。如果入射角大于临界角，光束将会反射。

真空的折射率被设定为1。通常，我们认定空气的折射率也为1（事实上要稍微大一点）。

水的折射率通常是1.33。玻璃的折射率在1.5左右，该值可以通过改变玻璃的组成成分来调整。

12.4.2 光纤的特性

要使得数据信号在光纤内部传输，首先要保证光束的入射角度大于临界角。如图 12-1 所示，光纤由三部分组成。中间的纤芯由非常纯净的玻璃组成，它的折射率为 1.5。纤芯的直径通常是在 50～125μm 之间。包围纤芯的玻璃，称为包层，有纯度稍差的玻璃组成，它的折射率为 1.45。纤芯或包层加在一起的直径在 125～440μm 之间。包层外部是一层保护层，该保护层称为护套，由柔性硅组成。



译图 12-1 在光纤中传输的光束

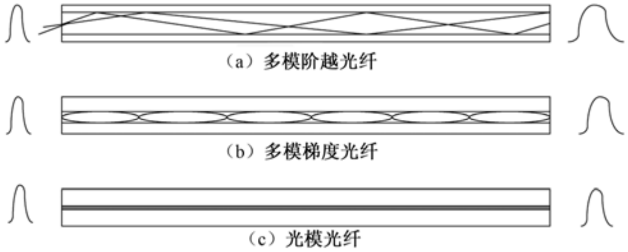
当光束导入到光纤的末端时，只要其入射角度大于临界角，该道光束将沿着光纤内部传输。光束接触到纤芯和包层的界面就会被反射回光纤。光纤的可接受角度由界面的临界角决定的。如果我们旋转这个可接受角度，就会产生一个锥形。只要光束的入射角度落在了这个锥形区域之内，光束就会沿着光纤传输。一旦光束被导入进光纤，它就会沿着纤芯不断的反弹前进，即在纤芯和包层的界面之间不断反射。

图12-1演示了光束是如何沿着光纤内部的介质界面反射前进的。如果纤芯的物理尺寸相对较大，各道光束以不同的入射角度进入光纤时，它们就会以不同的角度在光纤内部反射。因为它们在光纤内部的传输路径不一致，光束之间的传输距离也不尽相同。因此，各道光束将在不同的时间内到达接收端。脉冲信号在光纤内传输后，该脉冲在接收端就会出现展宽的现象，这使得信号的质量变差。这种现象称为模间色散。另一种会导致信号质量变差的现象称为色度色散。色度色散由不同波长的光束在光纤内部传输速度的差异形成的。当一串脉冲沿着光纤传输时，模间色散和色度色散会导致脉冲变的很宽而不可辨认，因而数据就会流失。

光纤的另一特性就是衰减。虽然用在光纤纤芯上的玻璃非常纯净，但不可能达到百分之百的纯净。因此，光束在光纤内部被吸收掉了。其他的信号损失包括弯曲、散射损耗和接续损耗。接续损耗是由光纤末端的偏移或者末端表面的不恰当抛光所导致的。

12.4.3 光纤的类型

光纤的生产类型有三种：多模阶越光纤、多模梯度光纤和单模光纤。多模阶越光纤的直径最大（通常在 50～100μm 之间）。更大的直径使得光束在光缆里反弹时传输的距离更长。多模光纤通常传输波长为 850～1300nm 的光信号。图 12-2 说明了窄脉冲在导入到光纤后，在接收端展宽的现象。



译图 12-2 窄脉冲导入光纤

图 12-2（a）的多模阶越光纤因为直径较大，所以相对来说更易于切割和端接。和其他类型的光纤相比，多模阶越光纤的生产成本更低。然而，它的传输速度太慢，因而在现代通信系统中的应用并不多。

图12-2（b）所示的多模梯度光纤是以这样一种方式组成的：在纤芯和包层之间的折射率是渐进变化的。这使得光束在光纤内部是渐进地弯曲前进的。相应的反射模式也显得更平缓，这样色散的程度被降低了。这带来了更好的性能，而在成本上只是适度的增加。梯度光纤比阶越光纤可以提供更大的带宽。

图12-2（c）中的单模光纤的性能在三种光纤中最高。它的直径很小（通常8μm），当光束导入到光纤时，折射保持在最小程度。光束近似于直线在纤芯中传输。脉冲在发送端被导入后，然后在接受端重现时，色散的程度很低。通常，单模光纤传输波长为1320nm或1550nm的光信号。单模光纤相对来说更贵一点。因为纤芯必须精确的重合，因为也更难于切割和端接。

相对于多模光纤，单模光纤提供更低的衰减特性。典型的单模光纤传输1310nm的光信号时每千米的衰减为 0.5dB。多模光纤传输同样的光信号，它的衰减达到了每千米3dB。单模光纤通常用在对带宽需求较高、长距离传输的场合。一些以太网光纤设备使用多模光纤可以传输2千米，而使用单模光纤时，可以传输70千米。

12.4.4 光缆的优势

抗噪性能

抗噪性能是光纤在工业应用中最出色的特性之一，光纤（的性能）在电磁干扰不可避免且强度很大的情况下（几乎）不受（任何）影响。光缆被起保护作用的护套包围，信号的传输沿着（保护套）里面的纤芯里进行。因此，没有必要把光缆和电缆隔离开，这使得光缆的路由配置更简单。

电气隔离

接地回路噪声和共模电位差的问题在光缆里是不存在的。设备产生的高电位场信号，可以被耦合进其他设备并转为低电位状态，而不会损坏设备。这在工业应用中非常有用。

低误码率

只要链路接收端的信号电平足够高，光纤系统的误码率就会非常低。

高危地区的高应用安全性

光纤链路可以在爆炸性环境中传输信号，这是因为其不具有足够的能量，从而不会引发爆炸。这一特性使光纤技术在设计高安全性系统时极其有用。

高带宽

光纤可以提供GHz级别的高带宽，大量低带宽信号可以被复用进同一条光缆（传输）。在商用（光传输）系统中，光缆可以用来传输各种类型的信号，比如语音、视频和数据。

低信号衰减

光纤由于吸收和散射的原因也会有信号衰减现象，但这种衰减与频率无关，其特性与铜质电缆（的衰减）完全不同。

质量轻、体积小

由于大量信号被复用进同一条光纤，光缆的体积和质量（相对）更小和更轻。这使得光缆的安装非常方便。

无串扰

因为光纤不积累电磁干扰，相邻光缆之间的信号不会发生串扰。

信号的安全性

对于要考虑信号安全问题的应用环境，光纤是非常优秀的解决方案。由于光缆不会产生电磁场而被外部的探测器检测到，相比于普通铜质线缆，从光纤中窃取信号的难度很高。

Unit 13 Voice over Internet Protocol

13.1 Text

Voice over Internet protocol - VoIP, or ***IP telephony*** is a technology by which the routing of voice communications are done through Internet or any other ***Internet Protocol*** **(*IP*)** based networks. Here the voice data is transmitted over a general purpose packet-switched network instead of dedicated traditional circuit-switched voice transmission lines.

VoIP is a part of the group of technologies called ***voice over packet networks***. Other network protocols like ***asynchronous transfer mode*** **(*ATM*)** can perform similar functions.

Though the concept of VoIP is simple, the implementation and applications of it is a bit complicated. In order to send voice, the information has to be separated into packets just like data. Packets are chunks of information broken up into the most efficient size for routing. From there, the packets need to be sent and put back together in an efficient manner. For more efficient use, the voice data can be compressed so that it require less space and will certainly record only a limited frequency range. There are many ways to compress audio, the algorithm for which is referred to as a **compressor/de-compressor (CODEC)**. Many a number of CODECs exist depending on the application (e.g., conversations, music, movies and sound recordings). The CODECs are optimized for compressing voice, which significantly reduce the bandwidth used compared to an uncompressed audio stream. ***Speech CODECs*** are optimized to improve spoken words at the expense of sounds outside the frequency range of human speech. Recorded music and other sounds do not generally sound very good when passed through a speech CODEC.

13.1.1 Protocols Used in VoIP Services

1．H.323

There are many protocols used in the implementation of VoIP services. The most popular VoIP signalling protocols are ***SIP*** and ***H.323***. Fundamentally, H.323 and SIP allow users to do the same thing: to establish multimedia communication (audio, video, or other data communication). However, H.323 and SIP differ significantly in design, with H.323 borrowing heavily from legacy communication systems and being a binary protocol, and with SIP not adopting many of the information elements found in legacy systems and being an ***ASCII-based protocol***. Supporters of each protocol have debated at length as to which approach is better and the results are certainly mixed.

H.323 is a standard that specifies the components, protocols and procedures that provide multimedia communication services: real-time audio, video, and data communications over packet networks, including Internet protocol (IP) based networks. H.323 is part of a family of ITU-T recommendations called H.32x that provides multimedia communication services over a variety of networks.

H.323 was originally created to provide a mechanism for transporting multimedia applications over LANs but it has rapidly evolved to address the growing needs of VoIP networks. One strength of H.323 was the relatively early availability of a set of standards, not only defining the basic call model, but in addition the supplementary services, needed to address business communication expectations. H.323 was the first VoIP standard to adopt the IETF standard RTP to transport audio and video over IP networks.

2. RTP——Real-Time Transport Protocol

The ***real-time transport protocol* (*RTP*)** provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video or simulation data, over multicast or unicast network services. Applications typically run RTP on top of UDP to make use of its multiplexing and checksum services; both protocols contribute parts of the transport protocol functionality. However, RTP may be used with other suitable underlying network or transport protocols.

RTP itself does not provide any mechanism to ensure timely delivery or provide other quality-of-service guarantees, but relies on lower-layer services to do so. It does not guarantee delivery or prevent out-of-order delivery, nor does it assume that the underlying network is reliable and delivers packets in sequence. The sequence numbers included in RTP allow the receiver to reconstruct the sender's packet sequence.

RTP consists of two closely-linked parts:

- The real-time transport protocol (RTP), to carry data that has real-time properties.
- The RTP control protocol (RTCP), to monitor the quality of service and to convey information about the participants in an on-going session.

3. SIP—Session Initiation Protocol

Session Initiation Protocol (SIP) is a protocol developed by IETF to assist in providing advanced telephony services across the Internet. Basically is a signalling protocol used for establishing sessions in an IP network. The ability to establish these sessions means that a host of innovative services become possible, such as IP Centrex services.

SIP is modeled upon other Internet protocols such as SMTP (Simple Mail Transfer Protocol) and HTTP (Hypertext Transfer Protocol.) It is used to establish, change and tear down (end) calls between one or more users in an IP-based network. In order to provide telephony services there is a need for a number of different standards and protocols to come together - specifically to ensure transport (RTP), signaling inter-working with today's telephony network, to be able to guarantee voice quality (RSVP, YESSIR), to be able to provide directories (LDAP), to authenticate users (RADIUS, DIAMETER), and to scale to meet the anticipated growth curves.

13.1.2 How does VoIP Work?

When you speak at the handset or a mike or a microphone, your voice generates electrical signals inside the gadget. These are analog signals i.e. the voltage level can take up any value within a range.

The analog signal is converted to a digital signal using an algorithm implemented by the device you are using. It can be a stand-alone VoIP phone or a ***softphone*** running on your PC. If you are using an analog phone, you will need a Telephony Adapter (TA) for this purpose. The digitized voice is arranged in packets (i.e. collection of bytes) and sent over the IP network.

The data is channeled through gateways and servers to the destination. If the called number is on the ***PSTN***, the server opens a connection to the PSTN and routes your call there.

While going to the PSTN or at the end device of a VoIP connection, the voice is again brought back to its analog form so that it is perceptible to a human ear.

During the entire process a protocol like SIP or H.323 is used to control the call (e.g. setting up connection, dialing, disconnecting etc.) and RTP is used for reliable transmission of data packets and maintain Quality of Service.

The digitization of analog voice signals is a must to transmit voice over the digital IP network. It can be done in several ways.

***PCM* (*Pulse Code Modulation*)** is a simple technique of sampling the sound signal at a fixed rate (8000 times/second) and generate a number corresponding to each sample. It assumes no specific property of the signal. So it works reasonably well with all types of sounds.

***LPC* (*Liner Predictive Coding*)** assumes specific properties of human voice and uses a more complex algorithm to digitize and compress voice data. It works well for sending human utterances offering a low data rate but is not suitable for transmitting music or fax.

***SBC* (*Sub Band Coder*)** uses a different approach of representing sounds in terms of frequencies rather than sampling at regular intervals.

Hybrid coders like the ***CELP* (*Code Excited Linear Prediction*)** use a mixture of the techniques to transmit sound of adequate quality.

13.1.3 Handoff Strategies

When a mobile moves into a different cell while a conversation is in progress, the MSC automatically transfers the call to a new channel belonging to the new base station. This ***handoff*** operation not only involves identifying a new base station, but also requires that the voice and control signals be allocated to channels associated within the new base station.

Many handoff strategies prioritize handoff requests over call initiation requests when allocating unused channels in a cell site. Handoff must be performed successfully and as infrequently as possible, and be imperceptible to the users. Once a particular signal level is specified as the minimum usable signal for acceptable voice quality at the base station receiver (normally taken as between −90 dBm and −100dBm), a slightly stronger signal level is used as a ***threshold*** at which a handoff is made. Figure13-1 (a) demonstrates the case where a handoff is not made and the signal drops below the minimum acceptable level to keep the channel active.

This dropped call event can happen when there is an excessive delay by the MSC in assigning a handoff or when the threshold Δ is set too small for the handoff time in the system. Excessive delays may occur during high traffic conditions due to computational loading at the MSC or due to the fact that no channels are available on any of the nearby base stations (thus forcing the MSC to wait until a channel in a nearby cell becomes free).

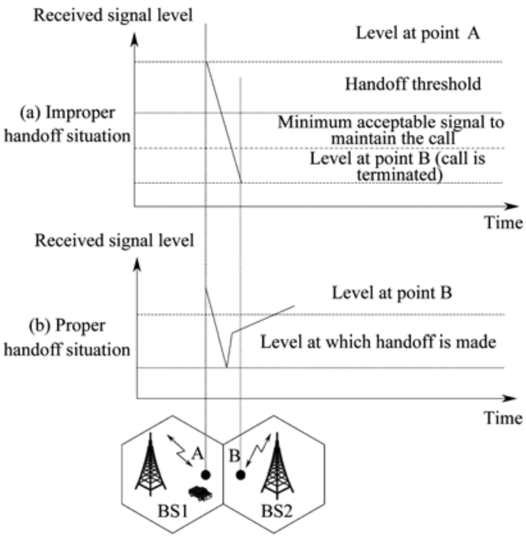


Figure 13-1 Handoff Process

The time over which a call may be maintained within a cell, without handoff, is called the ***dwell time***. The dwell time of a particular user is governed by a number of factors, including propagation, interference, distance between the subscriber and the base station, and other time varying effects.

In first generation analog cellular systems, signal strength measurements are made by the base stations and supervised by the MSC. Each base station constantly monitors the signal strengths of all of its reverse voice channels to determine the relative location of each mobile user with respect to the base station tower.

In today's second generation systems, handoff decisions are mobile assisted. In ***mobile assisted handoff*** (***MAHO***), every mobile station measures the received power from surrounding base stations and continually reports the results of these measurements to the serving base station. A handoff is initiated when the power received from the base station of a neighboring cell begins to exceed the power received from the current base station by a certain level or for a certain period of time. The MAHO method enables the call to be handed over between base stations at a much faster rate than in first generation analog systems since the handoff measurements are made by each mobile, and the MSC no longer constantly monitors signal handoff. MAHO is particularly suited for microcellular environments where handoffs are more frequent.

During the course of a call, if a mobile moves from one cellular system to a different cellular system controlled by a different MSC, an intersystem handoff becomes necessary. An MSC engages in an intersystem handoff when a mobile signal becomes weak in a given cell and the MSC cannot find another cell within its system to which it can transfer the call in progress.

Different systems have different policies and methods for managing handoff requests. Some systems handle handoff requests in the same way they handle originating calls.

13.1.4 VoIP on 3G will beat Wi-Fi

Mobile VoIP is set to grow, but it will run over the 3G data provided by cellular handsets, rather than over Wi-Fi, according to a research report from Disruptive Analysis Ltd., which predicts 250 million users of 3G VoIP by 2012, compared with less than 100 million for voice on Wi-Fi.

"Yes, 250 million is a surprisingly big number," said Dean Bubley of Disruptive Analysis. "I did the sums and then had to triple-check my own model. "The fundamental truth, he said, is that mobile networks are moving toward Long Term Evolution or possibly Ultra Mobile Broadband technology and that will be all IP based, so for operators "VoIP is mandatory."

VoIP will allow carriers to handle more calls on their scarce spectrum, and reduce expenses by handling all traffic as data. It will also let them offer new services such as push-to-talk and voice-integrated "mashups, " said Bubley, in his report, VoIP on 3G Business Models.

VoIP on 3G also fits the move to femtocells, which use the subscribers'broadband service to increase coverage in the home, since the digitized voice is ready to be handled by the femto's broadband backhaul.

Technical words and phrases

packet-switched 分组交换

dedicated ['dedikeitid] *a.* 专用的

general ['dʒenərəl] *a.* 一般的,普遍的

implementation [implimen'teiʃən] *n.* 安装启用，实行，履行

audio ['ɔːdiəu] *a.* 音频的，声音的

circuit-switched 电路交换

asynchronous transfer mode (ATM) 异步转移模式，异步传输模式

chunk [tʃʌŋk] *n.* 大块

adopt [ə'dɔpt] *v.* 采纳,采用；正式通过；领养

algorithm ['ælgəriðm] *n.* 算法

compressor [kəmpresə] *n.* 压缩器

de-compressor 解压缩器

legacy ['legəsi] *a.* 传统的

at length 最后，终于；详细地，充分地

interactive [intər'æktiv] *a.* 交互的，互相影响的，交互

transport [træns'pɔːt] *vt.* 传送，运输，流放

real-time transport protocol (RTP) 实时传输协议

checksum 校验和

quality-of-service 服务质量

mike [maik] *n.* 懒惰，游手好闲，扩音器，话筒

gadget ['gædʒit] *n.* 小配件，小玩意

stand-alone ['stændələun] *a.* （计算机的外围设备）能独立运行的

softphone 软电话

LPC (Liner Predictive Coding)　线性预测编码
adapter　[ə'dæptə]　*n.*　适配器
perceptible　[pə'septibl]　*a.*　可察觉的，能感觉得到的，看得见的
Sub Band Coder　子带编码器
CELP (Code Excited Linear Prediction)　码激励线性预测编码
IETF　互联网工程任务组
utterance　['ʌtərəns]　*n.*　说话，发表，说话的方式
supplementary　[sʌpli'mentəri]　*a.*　补足的，补充的，追加的
scale　[skeil]　*v.*　依比例决定
authenticate　[ɔ:'θentikeit]　*vt.*　证明，证实，鉴定
handoff　['hændɔf]　*n.*　切换
imperceptible　[impə'septəbl]　*a.*　不能感知的，不知不觉的
threshold　['θreʃəuld]　*n.*　阈值；门限，
excessive　[ik'sesiv]　*a.*　过度的，格外的，极端的
dwell time　驻留时间
propagation　[prɔpə'geiʃən]　*n.*　传播
reverse voice channel　方向语音信道
MAHO (mobile assisted handoff)　移动台辅助切换
intersystem　［计］系统间
Wi-Fi　WirelessFidelity　无线保真
Femtocell　毫微微蜂窝式基站

13.2　Reading Materials

1. Bluetooth and PANs

Bluetooth is an open standard that has been embraced by over 1000 manufacturers of electronic appliances. It provides an ad-hoc approach for enabling various devices to communicate with one another within a minimal 10 meter range. Named after King Harald Bluetooth, the 10th century Viking who united Denmark and Norway, the Bluetooth standard aims to unify the connectivity chores of appliances within the personal workspace of an individual.

Bluetooth operates in the 2.4 GHz ISM Band (2400～2483.5 MHz) and uses a frequency hopping TDD scheme for each radio channel. Each Bluetooth radio channel has a 1MHz bandwidth and hops at a rate of approximately 1600 hops per second. The frequency hopping scheme of each Bluetooth user is determined from a cyclic code of length $2^{27}-1$, and each user has a channel symbol rate of 1Mbit/s using GFSK modulation. The standard has been designed to support operation in very high interference levels and relies on a number of forward error control coding (FEC) and automatic repeat request (ARQ) schemes to support a raw channel bit error rate (BER) of about 10^{-3}.

Different countries have allocated various channels for Bluetooth operation. In the US and

most of Europe, the FHSS 2.4 GHz ISM band is available for Bluetooth use (see Table 13-1). A detailed list of states are defined in the Bluetooth standard to support a wide range of applications, appliances, and potential uses of the Personal Area Network. Audio, text, data, and even video is contemplated in the Bluetooth standard.

Table 13-1 IEEE 802.11b Channels for Both DS-SS and FH-SS WLAN Standards

Country	Frequency Range Available	DSSS Channels Available	FHSS Channels Available
United States	2.4～2.4835 GHz	1 through 11	2 through 80
Canada	2.4～2.4835 GHz	1 through 11	2 through 80
Japan	2.4～2.497 GHz	1 through 14	2 through 95
France	2.4465～2.4835 GHz	10 through 13	48 through 82
Spain	2.445～2.4835 GHz	10 through 11	47 through 73
Remainder of Europe	2.4～2.4835	1 through 13	2 through 80

The IEEE 802.15 standards committee has been formed to provide an international forum for developing Bluetooth and other PANs that interconnect pocket PCs, personal digital assistants (PDAs), cellphone, light projectors, and other appliances. With the rapid proliferation of wearable computers, such as PDAs, cellphones, smart cards, and position location devices, PANs may provide the connection to an entire new era of remote retrieval and monitoring of the world around us. PANs in Figure 13-2, as example according to Bluetooth standard, are provided by the standard.

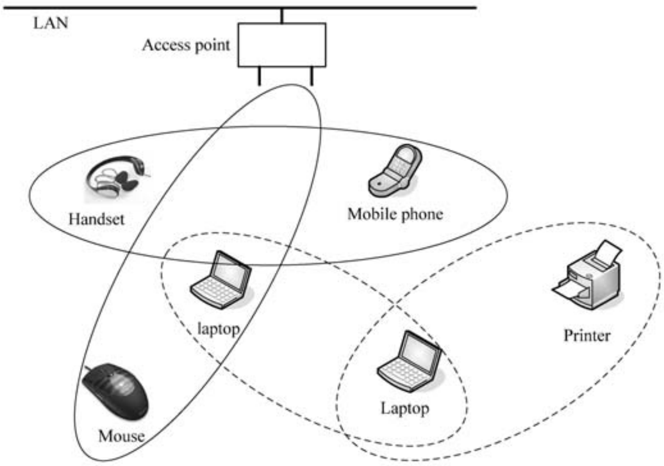


Figure 13-2 Example of PAN according to Bluetooth standard

2. KVH Migrates to Nortel VoIP NGN to Enable Network for Future Services Expansion

OCTOBER 12, 2006

TOKYO – KVH Co. Ltd., a leading provider of business communication and managed IT services in the Asia Pacific region, is migrating its voice infrastructure to an IP-based, next generation network (NGN) from Nortel to support future expansion of communication services such as hosted IP voice and unified messaging.

In order to meet the emerging needs of enterprise customers seeking more IP-based, value added services and to expand sales and improve efficiency, KVH is upgrading its current optical network infrastructure to a next generation IP network. The new network will be designed as a single integrated platform to simplify operations and offer expanded communications services which can be used by many types of terminals running a multitude of applications.

"Upgrading our voice network is the first step towards a full transition to IP," said Rakesh Bhasin, chairman, president and CEO, KVH. "Our customers'needs for highly flexible and value added services have been growing, and to meet those needs comprehensively we decided to migrate to a next generation IP-based network. We selected Nortel for VoIP NGN because of the carrier-grade quality of its communications systems and our experience, as a long-time Nortel customer, of the company's commitment to ongoing support."

"Nortel is a leader in Japan's growing momentum toward IP-based NGN networks to power advanced voice, data and multimedia services," said Nick Vreugdenhil, president, Japan, Nortel. "Our extensive NGN experience in North America, Europe and Asia is helping us demonstrate to Japanese customers how they can leverage our global experience and expertise to enhance competitiveness by enabling new revenue generating services, optimizing service delivery capabilities, and reducing operational costs."

At the core of the Nortel solution for KVH is the Nortel Communication Server 2000-Compact, a scalable, reduced-footprint, superclass softswitch supporting industry-standard VoIP protocols and providing an investment-protecting evolutionary path to a range of other services KVH is planning for the future. These plans include hosted IP business voice services and unified messaging to enable access to voicemail, e-mail or fax from any touchtone telephone or desktop PC.

3. Broadcom's Leading VoIP Solutions Now Available in TCL Communications Equipment VoIP Products

BOSTON, Fall 2005 VON Conference & Expo, Sept 20, 2005—Broadcom Corporation, a global leader in wired and wireless broadband communications semiconductors, and TCL Communications Equipment (Huizhou) Co., Ltd., the leading telephone manufacturer in China, today jointly announced that Broadcom's single-chip Voiceover Internet Protocol (VoIP) phone and analog terminal adapter (ATA) solutions are now shipping in TCL's consumer VoIP products. The election by TCL to use Broadcom® VoIP chips further exemplifies how Broadcom is growing its share of the worldwide VoIP market.

The two VoIP chips selected by TCL include the BCM1113R and the BCM1112. The BCM1113R is a low-cost IP phone chip that enables manufacturers to economically build feature-rich IP phones for the residential, small office/home office (SOHO), and small-to-medium sized business (SMB) markets. The BCM1112 is a low-cost VoIP CPE engine chip that enables manufacturers to build high-quality ATAs for residential deployments. ATAs connect legacy telephones to service providers' advanced IP communications networks. "Broadcom continues to strengthen its position in the VoIP market by partnering with the world's leading VoIP phone manufacturers and by offering high-quality, cost-effective VoIP products," said Patrick Sullivan,

Vice President and General Manager of Broadcom's VoIP phone products. “TCL's market-leadership, coupled with Broadcom's industry-leading VoIP portfolio, will enable high-quality IP phone and ATA products that will accelerate VoIP adoption in both the residential and SOHO markets worldwide.”

“The industry leading system-on-a-chip (SoC) functionality of Broadcom's BCM1113R and BCM1112 VoIP solutions has enabled us to develop high-quality VoIP products in a short design cycle with high stability and low risk," said Gu Gong, General Manager of TCL Communications. "We looked to Broadcom's VoIP expertise and market leadership for our VoIP products to ensure that TCL's strong product quality reputation was not compromised as we focus more on IP telephony products versus legacy analog telephone products.”

With Broadcom's VoIP chips and its field-proven xChange™ VoIP software suite, TCL has developed a cost-effective family of IP phones and residential ATA products that offer numerous features and functionality. Each of the new TCL IP phones (Models TCL-IP(8) and TCL-IP(9)) are based on the BCM1113R and feature session initiation protocol (SIP) and H.323 call signaling, hands-free full-duplex speakerphone capabilities, and standards-based voice codecs.

TCL's ATA (Model TCL-IAD501) is based on Broadcom's BCM1112 single-chip engine and features one analog POTS (plain old telephone set) port for connection to existing analog phones and fax machines. Both the IP phone and ATA products feature a Fast Ethernet switch and two physical Ethernet interfaces that support the connection between a broadband DSL or cable modem to a PC or local area network (LAN).

4. VoIP Impacting Security Appliance Market

SCOTTSDALE, Ariz., September 7, 2005—Though Voice over Internet Protocol (VoIP) adds many desirable new features to business communication systems, vendors and customers can face security challenges to realize these benefits, reports In-Stat (http://www.in-stat.com). As a result, more than 75% of the companies that have implemented VoIP plan to replace their security appliances within the next year, the high-tech market research firm says. The security appliance market is poised for strong growth over the next few years, and will reach $7 billion by 2009, In-Stat forecasts.

13.3 Exercises

1. Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）异步转移模式
（2）服务质量
（3）线性预测编码
（4）数据包
（5）二进制
（6）机制

（7）算法
（8）码激励线性预测编码
（9）蓝牙
（10）个人局域网
（11）切换
（12）软切换
（13）移动台辅助切换

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）Voice over Internet protocol is a technology by which the routing of voice communications are done through （　　　） or any other （　　　） based networks.
（2）In VoIP, the voice data is transmitted over a general purpose (　　　)-switched network instead of dedicated traditional (　　　)-switched voice transmission lines.
（3）In order to send voice, the information has to be separated into(　　　) just like data. Packets are chunks of information broken up into the most efficient size for (　　　).
（4）For more efficient use, the voice data can be (　　　) so that it require less space and will certainly record only a limited frequency range.
（5）The algorithm to compress audio is referred to as a (　　　).
（6）Speech CODECs are optimized to improve spoken words at the expense of sounds outside the (　　　) range of human speech. Recorded music and other sounds do not generally sound very good when passed through a (　　　) CODEC.
（8）The most popular VoIP signalling protocols used in the implementation of VoIP services are (　　　) and (　　　).
（9）The real-time transport protocol (RTP) provides (　　　) delivery services for data with (　　　) characteristics, such as interactive audio and video or simulation data, over multicast or unicast network services.
（10）PCM is a simple technique of sampling the sound signal at a fixed rate of (　　　) times/second and generate a (　　　) corresponding to each sample.
（11）Bluetooth operates in the （　　　） GHz ISM Band and uses a frequency hopping （　　　） scheme for each radio channel.
（12）Each Bluetooth radio channel has a (　　　) bandwidth and hops at a rate of approximately (　　　) hops per second.
（13）In order to support a raw channel bit error rate of about 10^{-3}, Bluetooth relies on (　　　) and (　　　) schemes.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）(　　) VoIP is a part of the group of technologies called voice over circuit networks.
（2）(　　) The bandwidth used is significantly reduced compared to an uncompressed audio stream.

（3）(　　) SIP borrows heavily from legacy communication systems and H.323 is an ASCII-based protocol.

（4）(　　) RTP may be used with other network or transport protocols.

（5）(　　) RTP itself does not provide any mechanism to ensure timely delivery or provide other quality-of-service guarantee.

（6）(　　) During the entire process RTP is used to control the call and SIP or H.323 is used for reliable transmission of data packets and maintain Quality of Service.

（7）(　　) H.323 is part of a family of ITU-T recommendations.

（8）(　　) Session Initiation Protocol is a protocol developed by ITU to assist in providing advanced telephony services across the Internet.

（9）(　　) The frequency hopping scheme of each Bluetooth user is determined from a cyclic code of length $2^{27}-1$.

（10）(　　) The handoff operation not only involves identifying a new base station, but also requires that the voice and control signals be allocated to channels associated within the new base station.

（11）(　　) Handoff strategies prioritize call initiation requests over handoff requests when allocating unused channels in a cell site.

（12）(　　) In first generation analog cellular systems, signal strength measurements are made by the mobile stations and supervised by the base stations.

（13）(　　) In the second generation systems, mobile station assists handoff decisions.

（14）(　　) In MAHO, a handoff is initiated when the power received from the base station of a neighboring cell begins to exceed the power received from the current base station by a certain level or for a certain period of time.

（15）(　　) MAHO is not suited for microcellular environments where handoffs are more frequent.

13.4　课文参考译文　基于 IP 的语音传输

互联网语音传输协议（VoIP 或 IP 电话），就是通过互联网或其他基于 IP 协议的网络进行语音信号的传输。其中，语音数据通过一个通用分组交换网来传输，而非传统的基于电路交换的语音传输线路。

VoIP 属于分组语音传输技术，其他如基于异步转移模式（ATM）的网络协议也可以实现类似的功能。

虽然 VoIP 的概念简单，其实现和应用却有一定的难度。语音信号在发送前必须被转换成数据包的形式，该数据包是（语音）信息块，被划分成适宜于路由、发送的大小，由接收端在接收后将其重新组合起来。为了提高带宽利用率，语音数据（信号在发送前）将被压缩，以占用、覆盖更少的频段。实现语音信号压缩的方式（或算法）有多种，（常将其）称为编码器/解码器（CODEC），对不同算法的选择主要取决于（相应的）应用（数据的类型），如会话、音乐、电影和录音。相比原始音频数据流，通过 CODEC 压缩后的语音信号受到优化，带宽明显降低。该优化针对的是语音信号中的词句，其代价是滤掉了人类音频之外的信号分量。因此，录制的音乐和其他声音信号通过上述语音 CODEC 后的效果

并不理想。

13.4.1　VoIP 中的协议

1．H.323 协议

有很多协议可以用来实现 VoIP 业务，其中最常用的 VoIP 协议就是 SIP 和 H.323，两者的功能都是建立多媒体通信业务（包括音频、视频或数据业务）。然而，H.323 和 SIP 在构架上差异很大。H.323 主要借鉴了传统的通信系统，是二进制协议。SIP 并没有采用传统系统的构架，是一个基于 ASCII 的协议。（尽管）两个协议的支持者都认为自己（支持）的协议具有更多的优势，但（实际上）两个协议各有特点。

H.323 标准定义了多媒体通信服务的组件、协议和程序，这些服务是包括 IP 网在内的、基于分组交换网的实时音频、视频和数据通信。H.323 是 ITU-T 推荐建议中 H.32x 家族的一部分，H.32x 提供各种网络上的多媒体业务。

H.323 原先是为局域网中传输多媒体而制定的规范，但现在它已经用在不断增长的 VoIP 网络中。H.323 的优势是一系列可用的规范，不仅仅定义了基本的呼叫模型，也定义了商务应用中需要的增值业务。H.323 是第一个采用 IETF 的 RTP 协议来传输 IP 网络音频和视频信号的 VoIP 规范。

2．实时传输协议（RTP）

实时传输协议为实时数据提供端到端的传输服务，如在多播或单播网络业务中的互动音频、视频或模拟数据。运行 RTP 的应用程序以 UDP 为基础，以便利用其复用和校验服务来实现数据传输的部分功能，但 RTP 协议也可以用在其他网络环境或传输协议上。

RTP 本身并没有提供任何机制以实现消息的准时投递，也不保证 QoS 质量，但是，它可以依赖底层的服务来实现（质量保证）。它不保证消息的可靠投递，也不预防消息会出现乱序状态。（同样地，）它也不保证网络的可靠性和数据包序列的先后顺序。数据包的序列号包含在 RTP 内部，这样，接收端可以重新对数据包进行排序。

RTP 由两层紧密相连的部分组成：

- 实时传输协议（RTP），传输实时数据。
- RTP 控制协议（RTCP），监控服务质量，在通话中发送有关通话方的信息。

3．会话初始协议（SIP）

会话初始协议（SIP）是由互联网工程任务组（IETF）开发的协议，用于提供基于互联网的高级电话业务。SIP 是一种信令协议，用于在 IP 网中建立通话连接，（其出现）预示着大量新型业务（的诞生）成为可能，如 IP Centrex 业务。

SIP 是以像 SMTP（简单邮件传输协议）和 HTTP（超本文传输协议）等互联网协议为蓝本制定的，用于在 IP 网中的一个或多个用户之间建立、更改和拆除呼叫（连接）。为了提供这些电话业务，有必要制定一系列不同的标准和协议——用于确保可靠传输（RTP），与现今电话网的信令互通，保证服务质量（RSVP、YESSIR），提供目录（LDAP），验证用户（RADIUS, DIAMETER），并根据市场的需求增长来调整。

13.4.2 VoIP 是如何运作的?

当你在手机、麦克风等扩音设备前讲话时，你的语音在这些装置里产生电信号。这是模拟信号，即电压是一定范围内的任意值。

模拟信号通过一定算法转换成数字信号，这是通过你所使用的装置来完成的。它可能是一部单独的 VoIP 电话或你计算机上的软电话。如果你使用一部模拟电话机，你需要一个电话适配器（TA）来完成这个转换。数字语音以数据包的形式发送到 IP 网络上。

数据穿透网关和服务器到达目的地。如果被叫号码在 PSTN 网里，服务器将建立到达 PSTN 的连接，并将你的呼叫路由到目的终端。

当进入 PSTN 网络或 VoIP 网络中的终端时，语音被重新转换成模拟信号，这样，人的耳朵就可以听到。

在整个过程中，像 SIP 或 H.323 的协议用来控制呼叫（比如建立连接、拨号、断开连接等），而 RTP 用来实现数据包的可靠传输和服务质量。

模拟语音信号的数字化是在数字 IP 网络上传输语音信号的一个必要条件。转化可以用几种方式实现。

PCM（脉冲振幅调制）是一项简易的技术，该技术以固定的速率（每秒 8000 次）采样声音信号，为每一个采样生成一个相应的数字。PCM 保证了信号的特定属性。因此，PCM 可以较好地采样各种信号。

LPC（线性预测编码）保证了人类语音的特定属性，并使用一种复杂的算法对语音数据进行数字化和压缩。在以较低的数据速率发送人类的语音信号时效果很好，但不适于传输音乐或传真。

SBC（子带编码器）使用一种不同的方法来标识声音信号，它根据频率而非等间隔的信号采样来标识声音信号。

混合调制器如 CELP（码激励线性预测码）使用混合技术来传送质量较好的声音信号。

13.4.3 切换策略

当移动台在通话期间从一个基站的覆盖区域移到另一个基站覆盖区域时，基站控制器（MSC）自动地将该通话转移到另一个基站的语音信道上去。这个切换过程不仅要识别新的基站，而且还要将语音和控制信号转移到新基站的信道上。

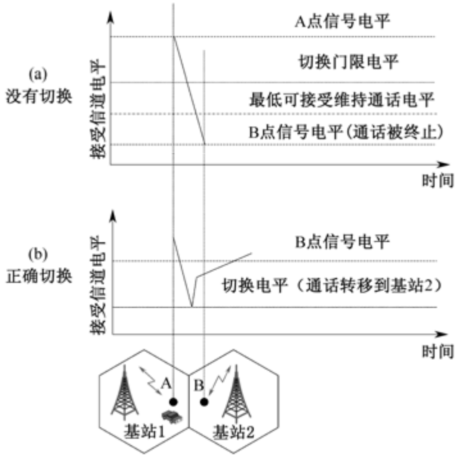
在为移动台分配空闲的信道时，许多切换策略都使切换请求优先于呼叫请求。切换过程必须顺利实现，且切换的次数越少越好，也不应被用户察觉。一旦某个信号电平被指定为基站接收机可接受通话质量的最小可用电平（通常在−90～−100dBm 之间），就可以设（比之）稍强一点的信号电平为切换门限电平。图 13-1（a）所示为没有采取切换，信号小于最低可接受电平的情况。

在基站控制器分配切换任务时延迟过大，或者对于系统切换时间来说，门限 Δ 设置得过小，都可能导致掉话。当基站控制器的话务量达满负荷时，或相邻基站没有可用信道时（这种情况下，基站控制器必须等待相邻小区腾出空闲的信道），会出现延迟过大的情况。

呼叫在一个小区内没有经过切换的通话时间称为驻留时间。每个用户的驻留时间由很多因素来决定，包括传播特性、干扰、用户与基站之间的距离和其他一些时变条件。

在第一代模拟蜂窝系统中，由基站来测量信号强度，而基站控制器则监视信号电平。每个基站频繁地监视它所有反向语音信道中的信号强度，以测算移动用户相对于基

站的位置。



译图 13-1　小区边界的越区切换过程示意图

在今天的第二代移动通信系统中，切换在移动台的辅助下进行。移动台辅助切换（MAHO），即每个移动台测量周围基站的信号强度，并将测量数据连续地汇报给当前为它服务的基站。当从一个相邻小区的基站中接收到的信号强度比当前基站的信号强度高出一定电平时，或是持续了一定的时间时，就要进行切换。移动台辅助切换方法使得通话在基站间的切换比在第一代模拟系统中快得多，因为切换的检测是由每个移动台来完成的，因而基站控制器就不再需要连续地监视信号强度。移动台辅助切换在需要频繁切换的微蜂窝环境下特别适用。

在一个通话过程中，如果移动台从一个蜂窝系统转移到另一个由不同基站控制器（MSC）控制的蜂窝系统中，则需要进行系统间的切换。当小区中移动台的信号强度减弱，同时基站控制器在它自己的系统中找不到一个小区来转移正在进行的通话时，该基站控制器必须进行系统间切换。

不一样的系统用不同的策略和方法来处理切换请求。某些系统处理切换请求的方式与处理呼叫请求是一样的。

13.4.4　3G VoIP 将击败 Wi-Fi

英国电信咨询公司 Disruptive Analysis 在一份研究报告中指出：处于增长态势的移动 VoIP 技术将运行在 3G 而非 Wi-Fi 上。它还进一步预测：到 2012 年，3G VoIP 的用户数量将超过 2.5 亿，而语音 Wi-Fi 的用户数则不到 1 亿。

“是的，2.5 亿可是个大数目！”Disruptive Analysis 公司的 Dean Bubley 证实道，“我统计了总数，然后不得不对统计模型进行了三次核查。”他还解释说，由于移动网正向“长期演进计划”或“终极移动宽带”发展过渡，而这些都是基于 IP 的，因此，对运营商来说“必须采用 VoIP 技术”。

Bubley 还在他的报告《VoIP 3G 商务模型》中说：通过将所有通信内容以数据方式进行传输，VoIP 可以使运营商在有限的带宽资源上实现更多的电话呼叫，并降低成本。VoIP 还支持运营商推出更多的新业务，如“一键通”和集成语音的“mashup”业务。

Unit 14 ARM Technique and Embedded Systems

14.1 Text

14.1.1 ARM Technique

The ARM architecture describes a family of RISC-based computer processors designed and licensed by British company ARM Holdings. It was first developed in the 1980s and globally as of 2013 is the most widely used 32-bit instruction set architecture in terms of quantity produced. In 2011 alone, producers of chips based on ARM architectures reported shipments of 7.9 billion ARM-based processors, representing 95% of smartphones, 90% of hard disk drives, 40% of digital televisions and set-top boxes, 15% of microcontrollers and 20% of mobile computers.

As an IP core business, ARM Holdings itself does not manufacture its own electronic chips, but licenses its designs to other semiconductor manufacturers. ARM-based processors and systems on a chip include the Qualcomm Snapdragon, nVidia Tegra, and Texas Instruments OMAP, as well as ARM's Cortex series and Apple System on Chips (used in its iPhones). The name was originally an acronym for Advanced RISC Machine, and in its early days Acorn RISC Machine.

Figure 14-1 The ARM logo

Using a RISC based approach to computer design, ARM processors require significantly fewer transistors than processors that would typically be found in a traditional computer. The benefits of this approach are lower costs, less heat, and less power usage, traits that are desirable for use in light, portable, battery-powered devices such as smart phones and tablet computers. The reduced complexity and simpler design allows companies to build a low-energy system on a chip for an embedded system incorporating memory, interfaces, radios, etc. The earliest example was the Apple Newton tablet but this same approach is still used in the Apple A4 and A5 chips in the iPad.

ARM periodically releases updates to its core—currently ARMv7 and ARMv8—which chip manufacturers can then license and use for their own devices. Variants are available for each of these to include or exclude optional capabilities. More recently, ARM architecture has included 64-bit versions - in 2012, Microsoft produced its new Surface RT tablet with ARM technology and AMD announced that it would start producing server chips based on the 64-bit ARM core in 2014.

Figure 14-2 A Conexant ARM processor used mainly in routers

ARM offers several microprocessor core designs, including the ARM7, ARM9, ARM11, Cortex-A8, Cortex-A9, and Cortex-A15. Companies often license these designs from

ARM to manufacture and integrate into their own system on a chip (SoC) with other components like RAM, GPUs, or radio basebands (for mobile phones).

ARM offers a variety of licensing terms. To all licensees, ARM provides an integratable hardware description of the ARM core, as well as complete software development toolset (compiler, debugger, SDK), and the right to sell manufactured silicon containing the ARM CPU.

While ARM does not grant the licensee the right to resell the ARM architecture itself, licensees may freely sell manufactured product (chip devices, evaluation boards, complete systems, etc.). Merchant foundries can be a special case; not only are they allowed to sell finished silicon containing ARM cores, they generally hold the right to re-manufacture ARM cores for other customers.

ARM's 2006 annual report and accounts state that royalties totalling £88.7 million ($164.1 million) were the result of licensees shipping 2.45 billion units. This is equivalent to £0.036 ($0.067) per unit shipped. This is averaged across all cores, including expensive new cores and inexpensive older cores.

In the same year ARM’s licensing revenues for processor cores were £65.2 million (US$119.5 million), in a year when 65 processor licenses were signed, an average of £1 million ($1.84 million) per license. Again, this is averaged across both new and old cores.

Given that ARM's 2006 income from processor cores was approximately 60% from royalties and 40% from licenses, ARM makes the equivalent of £0.06 ($0.11) per unit shipped including both royalties and licenses.

Table 14-1　ARM cores

Architecture	Family
ARMv1	ARM1
ARMv2	ARM2, ARM3, Amber
ARMv3	ARM6, ARM7
ARMv4	StrongARM, ARM7TDMI, ARM8, ARM9TDMI, FA526
ARMv5	ARM7EJ, ARM9E, ARM10E, XScale, FA626TE, Feroceon, PJ1/Mohawk
ARMv6	ARM11
ARMv6-M	ARM Cortex-M0, ARM Cortex-M0+, ARM Cortex-M1
ARMv7	ARM Cortex-A5, ARM Cortex-A7, ARM Cortex-A8, ARM Cortex-A9, ARM Cortex-A15, ARM Cortex-R4, ARM Cortex-R5, ARM Cortex-R7, Scorpion, Krait, PJ4/Sheeva, Swift
ARMv7-M	ARM Cortex-M3, ARM Cortex-M4
ARMv8-A	ARM Cortex-A53, ARM Cortex-A57 , X-Gene

14.1.2　Embedded Systems

An embedded system is a computer system designed for specific control functions within a larger system. It is embedded as part of a complete device often including hardware and mechanical parts. By contrast, a general-purpose computer, such as a personal computer (PC), is designed to be flexible and to meet a wide range of end-user needs. Embedded systems control many devices in common use today.

Embedded systems contain processing cores that are typically either microcontrollers-or digital signal processors (DSP). The key characteristic, however, is being dedicated to handle a particular task. Since the embedded system is dedicated to specific tasks, design engineers can optimize it to reduce the size and cost of the product and increase the reliability and performance. Some embedded systems are mass-produced, benefiting from economies of scale.

Physically, embedded systems range from portable devices such as digital watches and MP3 players, to large stationary installations like traffic lights, factory controllers, or the systems controlling nuclear power plants. Complexity varies from low, with a single microcontroller chip, to very high with multiple units, peripherals and networks mounted inside a large chassis or enclosure.

Embedded systems span all aspects of modern life and there are many examples of their use. Telecommunications systems employ numerous embedded systems from telephone switches for the network to mobile phones at the end-user. Computer networking uses dedicated routers and network bridges to route data.

Consumer electronics include personal digital assistants (PDAs), mp3 players, mobile phones, digital cameras, DVD players, GPS receivers, and printers. Many household appliances, such as microwave ovens, washing machines and dishwashers, are including embedded systems to provide flexibility, efficiency and features.

Embedded systems are designed to do some specific task, rather than be a general-purpose computer for multiple tasks. Some also have real-time performance constraints that must be met, for reasons such as safety; others may have low or no performance requirements, allowing the system hardware to be simplified to reduce costs.

Many embedded systems consist of small, computerized parts within a larger device that serves a more general purpose. An embedded system in an automobile provides a specific function as a subsystem of the car itself.

The program instructions written for embedded systems are referred to as firmware, and are stored in read-only memory or Flash memory chips. They run with limited computer hardware resources: little memory, small or non-existent keyboard or screen.

Embedded systems range from no user interface at all — dedicated only to one task — to complex graphical user interfaces that resemble modern computer desktop operating systems. Simple embedded devices use buttons, LEDs, graphic or character LCDs with a simple menu system.

Embedded processors can be broken into two broad categories: ordinary microprocessors (μP) and microcontrollers (μC), which have many more peripherals on chip, reducing cost and size. Contrasting to the personal computer and server markets, a fairly large number of basic CPU architectures are used; there are Von Neumann as well as various degrees of Harvard architectures, RISC as well as non-RISC and VLIW.

Technical words and phrases

RISC [risk] 精简指令集计算机
instruction set 指令集

architecture ['ɑ:kitektʃə] *n.* 体系结构
chip [tʃip] *n.* 芯片
smartphone [sma:tfəun] *n.* 智能手机
hard disk drive 硬盘驱动器
set-top box 机顶盒
microcontroller [ˌmaikrəkən'trəulə] *n.* 微控制器
IP (Intellectual Property) 知识产权
semiconductor [ˌsemikən'dʌktə] *n.* ［物］半导体
Qualcomm *n.* 美国高通公司
Texas Instrument 德州仪器
transistor [træn'sistə] *n.* 晶体管
tablet computer 平板电脑
microprocessor ['maikrəuˌprɔsesə] *n.* [计]微处理器
SoC (System on Chip) 片上系统
RAM(Random Access Memory) 随机存取存储器
GPU(Graphic Processing Unit) 图形处理器
radio baseband 无线基带
licensee [ˌlaisən'si:] *n.* 持牌人；执照持有者；获许可的人
silicon ['silikən] *n.* 硅
evaluation board 评估板
foundry ['faundri:] *n.* 晶原代工厂
royalty ['rɔiəlti] *n.* 版税
general-purpose *adj.* 多方面的，多种用途的
digital signal processors 数字信号处理器
economies of scale 规模经济
peripheral [pə'rifərəl] *n.* 外围设备
telephone switch 电话交换机
router ['rautə] *n.* 路由器
network bridge 网桥
personal digital assistants (PDA) 个人数字助理
real-time ['ri:əlˌtaim] *adj.* 实时的
firmware ['fə:mwɛə] *n.* 固件
flash memory 闪存
Von Neumann 冯·诺依曼
VLIW(very long instruction word) 超长指令字

14.2 Reading Material

1. Smart phone

A smartphone is a mobile phone built on a mobile operating system, with more advanced

computing capability and connectivity than a feature phone. The first smartphones combined the functions of a personal digital assistant (PDA) with a mobile phone. Later models added the functionality of portable media players, low-end compact digital cameras, pocket video cameras, and GPS navigation units to form one multi-use device. Many modern smartphones also include high-resolution touchscreens and web browsers that display standard web pages as well as mobile-optimized sites. High-speed data access is provided by Wi-Fi and mobile broadband. In recent years, the rapid development of mobile app markets and of mobile commerce have been drivers of smartphone adoption.

2. Android

Android is an open-source platform founded in October 2003 by Andy Rubin and backed by Google, along with major hardware and software developers (such as Intel, HTC, ARM, Motorola and Samsung, to name a few), that form the Open Handset Alliance. The first phone to use Android was the HTC Dream, branded for distribution by T-Mobile as the G1. The software suite included on the phone consists of integration with Google's proprietary applications, such as Maps, Calendar, and Gmail, and a full HTML web browser. Android supports the execution of native applications and a preemptive multitasking capability. Third-party free and paid apps are available via Google Play, which launched in October 2008 as Android Market.

In January 2010, Google launched the Nexus One smartphone using its Android OS. Although Android has multi-touch abilities, Google initially removed that feature from the Nexus One, but it was added through a firmware update on February 2, 2010.

3. iOS

In 2007, Apple Inc. introduced the original iPhone, one of the first mobile phones to use a multi-touch interface. The iPhone was notable for its use of a large touchscreen for direct finger input as its main means of interaction, instead of a stylus, keyboard, and/or keypad as typical for smartphones at the time. It initially lacked the capability to install native applications, meaning some did not regard it as a smartphone. However in June 2007 Apple announced that the iPhone would support third-party “web 2.0 applications” running in its web browser that share the look and feel of the iPhone interface. A process called jailbreaking emerged quickly to provide unofficial third-party native applications to replace the built-in functions (such as a GPS unit, kitchen timer, radio, map book, calendar, notepad, and many others).

14.3 Exercises

1．Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）指令集
（2）机顶盒
（3）微控制器

（4）半导体
（5）晶体管
（6）片上系统
（7）图形处理器
（8）无线基带
（9）数字信号处理
（10）外设
（11）固件
（12）超长指令字

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）The ARM architecture describes a family of（　　）-based computer processors designed and licensed by British company （　　） Holdings
（2）ARM was originally an acronym for (　　) RISC Machine, and in its early days (　　) RISC Machine
（3）The benefits of RISC-based ARM processors are lower (　　), less (　　), and less (　　), traits that are desirable for use in light, portable, (　　)-powered devices such as smart phones and tablet computers.
（4）Companies often acquire license from ARM to manufacture and integrate into their own system on a (　　) with other components like (　　), (　　)s, or (　　).
（5）ARM Ltd does not manufacture or sell CPU devices based on its own designs, but rather, licenses the (　　) architecture to interested parties.
（6）An embedded system is a computer system designed for specific (　　) functions within a larger system.
（7）Embedded systems contain processing cores that are typically either (　　) or (　　).
（8）The program instructions written for embedded systems are referred to as (　　), and are stored in (　　) memory or (　　) memory chips.
（9）Embedded processors can be broken into two broad categories: ordinary (　　) and (　　), which have many more (　　) on chip, reducing cost and size.
（10）An embedded system in an automobile provides a specific function as a (　　) of the car itself.

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）(　　) As an IP core business, ARM Holdings manufactures its own electronic chips.
（2）(　　) ARM-based processors and systems on a chip include the Qualcomm Snapdragon, nVidia Tegra, and Texas Instruments OMAP and so on.
（3）(　　) Compared with processors in traditional computer, ARM processors require significantly fewer transistors than processors.
（4）(　　) ARM architecture has included 64-bit versions as early as 1990s.

（5）(　　) Cortex-A8, Cortex-A9, and Cortex-A15 are categorized as state-of-the-art ARM processors.

（6）(　　) Merchant foundries generally do not hold the right to re-manufacture ARM cores for other customers.

（7）(　　) A general-purpose computer is designed to be flexible and to meet a wide range of end-user needs.

（8）(　　) The embedded system is dedicated to general-purpose tasks, it is not possible to reduce the size and cost of the product and increase the reliability and performance.

（9）(　　) Telephone switches for the network in the field of Telecommunication are embedded systems.

（10）(　　) All embedded systems have user interface, buttons, LEDs, graphic or character LCDs with a simple menu system.

（11）(　　) A fairly large number of basic CPU architectures are used in embedded systems such as Von Neumann, various degrees of Harvard architectures, RISC, non-RISC and VLIW.

14.4　课文参考译文　ARM 技术与嵌入式系统

14.4.1　ARM 技术

ARM 体系是指基于 RISC 技术的计算机处理器，由英国 ARM 公司设计并授权各生产厂商。ARM 技术最早可以追溯至 20 世纪 80 年代，截至 2013 年，它是世界上普及率最高的 32 位处理器。仅在 2011 年，ARM 芯片的出货量就达到了 79 亿片。据统计，95%的智能手机、90%的硬盘驱动器、40%的数字电视及机顶盒、15%的微控制器和 20%移动计算机都使用了 ARM 芯片。

作为一家出售内核设计的公司，ARM 本身并不生产 CPU，而是将其内核设计授权给其他半导体厂商进行生产。世界上优秀的 ARM 处理器有高通公司的骁龙系列、英伟达的图睿系列、德州仪器的 OMAP 系列、基于 Cortex 内核的 CPU 以及 iPhone 中的 Apple 系列处理器。ARM 是 Advanced RISC Machine（先进的精简指令集机器）三个单词的首字母缩写，但起初被称作 Acorn RISC Machine。

译图 14-1　ARM 标志

与传统 CPU 不同，ARM 处理器基于 RISC 技术，因而其内部集成的晶体管数量大大减少。这带来了低成本、低发热量以及低功耗的特性。这些特性对于智能手机或平板电脑等需要电池供电的移动设备来说是难能可贵的。由于 ARM 处理器的电路复杂度降低、设计更为精简，使得厂商可以生产出功耗极低的片上系统，即集成存储器、IO 接口、射频芯片等部件的嵌入式系统，如苹果公司早先推出的“牛顿”（Newton）掌上电脑。目前 iPad 内部使用的苹果 A4 和 A5 处理器，其集成度也非常高。

ARM 内核每隔数年进行升级换代，目前为 ARM v7 和 ARM v8 构架。半导体厂商在获取内核授权后就可以设计符合自身需要的 ARM 处理器，并可根据实际需求增设或简化、去除某些可选功能。最近，ARM 处理器推出了 64 位版本。2012 年，微软发布的 Surface RT 平板电脑也是基于 ARM 技术，AMD 公司宣布将在 2014 年推出面向服务器领域的 64 位 ARM 处

理器。

ARM 公司已推出的各类内核有 ARM7, ARM9, ARM11, Cortex-A8, Cortex-A9, 和 Cortex-A15。厂商在得到 ARM 公司授权后将 ARM 内核融合进自家的片上系统（SoC），如增加 RAM（随机访问存储器）、GPU（图形处理器）或无线基带芯片（主要用于手机通信领域）。

ARM 公司将处理器的构架授权给相关半导体设计厂商的形式灵活多样，其授权提供的相关内容如下：ARM 内核的硬件描述、完整的软件开发平台（编译器、调试器和软件开发包）、以及生产基于 ARM 内核半导体芯片的授权许可。

但是，ARM 公司不允许厂家再将 ARM 构架授权给第三方使用。得到授权的厂家可以出售其产品（芯片设备、评估开发板、完整的嵌入式系统等）。半导体代工厂则属特例，他们不但可以出售融合 ARM 内核的半导体芯片成品，也有权为其他客户提供代工服务。

译图 14-2　在路由器中广泛使用的美国科胜讯公司 ARM 处理器

ARM 公司 2006 年度财报显示，其版税金额达到 8870 万英镑（约合 1.641 亿美元），共计授权 24.5 亿片 ARM 处理器。折算下来每块 ARM 处理器的版税金额为 0.036 英镑（约合 0.067 美元）。当然这只是平均数，实际上新内核的费用将高些，而老产品则比较便宜。

同一年，ARM 公司在处理器授权领域获利达 6520 万英镑（约合 1.195 亿美元）。2006 年 ARM 签署了 65 份授权协议，每份授权约 100 万英镑（约合 184 万美元），当然这也是平均费用。

ARM 公司 2006 年处理器的收入中约 60%为版税金，40%为授权费。平摊下来每片 ARM 处理器约 0.06 英镑（约合 0.11 美元）。

译表 14-1　ARM 内核

体系构架	处理器
ARMv1	ARM1
ARMv2	ARM2, ARM3, Amber
ARMv3	ARM6, ARM7
ARMv4	StrongARM, ARM7TDMI, ARM8, ARM9TDMI, FA526
ARMv5	ARM7EJ, ARM9E, ARM10E, XScale, FA626TE, Feroceon, PJ1/Mohawk
ARMv6	ARM11
ARMv6-M	ARM Cortex-M0, ARM Cortex-M0+, ARM Cortex-M1
ARMv7	ARM Cortex-A5, ARM Cortex-A7, ARM Cortex-A8, ARM Cortex-A9, ARM Cortex-A15, ARM Cortex-R4, ARM Cortex-R5, ARM Cortex-R7, Scorpion, Krait, PJ4/Sheeva, Swift
ARMv7-M	ARM Cortex-M3, ARM Cortex-M4
ARMv8-A	ARM Cortex-A53, ARM Cortex-A57 , X-Gene

14.4.2　嵌入式系统

嵌入式系统为特定应用而设计，主要提供控制功能，通常嵌入在受控设备内部。与嵌入式系统截然不同的是，通用计算机系统，如个人电脑（PC）可以满足多种任务和各类终端用

户的需求，而嵌入式系统在各类设备中实现控制功能。

嵌入式系统内部集成了处理器内核，如微控制器或数字信号处理器（DSP），主要用于实现特定任务和目的。由于嵌入式系统的这一特征，设计人员可以对其进行优化从而减小系统的尺寸并降低成本，进而提高其可靠性和性能。此外，嵌入式系统还可通过量产以降低成本。

典型的嵌入式系统有数字手表或 MP3 播放器等便携设备，也有交通灯、核电站控制系统等大型设备。从系统集成度而言，嵌入式系统既可以是小型单片机，也可以是大型设备，其内部集成了各类功能模块、外设单元或网络。

嵌入式系统已渗入现代人类生活的各个角落。电信设备中集成了大量的嵌入式系统，如电话交换机。计算机通信网中用于数据转发的路由器和网桥也属于嵌入式系统。

消费电子类产品中的嵌入式设备有个人数字助理（PDA）、MP3 播放器、手机、数码相机、DVD 播放器、GPS 接收器和打印机等。众多的家用电器，如微波炉、洗衣机和洗碗机等内部均整合了嵌入式系统，充分利用其灵活、高效的特点。

不同于通用计算机功能较为全面，嵌入式系统用于实现特定任务。某些嵌入式系统必须具备实时处理的特性，以满足安全性的要求。在某些对性能无特别要求的场合下，可以对硬件进行精简以降低成本。

不少嵌入式系统由多个子系统组成，通过整合实现最终功能。汽车中的嵌入式系统实际上可视作汽车内部的一个子系统。

为嵌入式系统编写的程序也称为固件，存储在只读存储器或闪存芯片中。这类程序的运行对硬件资源需求相当小，通常仅占用很少的内存空间，甚至不需要键盘或屏幕。

有些嵌入式系统连人机交互界面都没有，这类系统应用较为特殊。而有些具备复杂的用户图形界面，与普通计算机的操作系统外观无异。一般嵌入式设备集成了按键、LED 灯等。LCD 液晶屏有只能显示字符的，也有能显示彩色图像的，液晶屏通常可以显示简易的用户菜单。

嵌入式系统中的处理器主要分为两大类：微处理器和微控制器，而微控制器内部集成的外设部件较多，这样可以降低整个系统的成本和体积。与个人电脑和服务器市场形成鲜明对比的是，嵌入式领域的 CPU 构架种类众多，有冯・诺依曼型，也有哈佛构架，有 RISC 也有非 RISC 或 VLIW 构架的。

Unit 15　Commonsensible Concepts

15.1　Text

1. ***C Band*** means a portion of the electromagnetic spectrum commonly used for both terrestrial and satellite radio transmission. In terrestrial microwave applications, the C Band is in the range of 4～6 GHz. Satellites split the C Band into an uplink channel of 5.925～6.425GHz and a downlink channel of 3.7～4.2GHz. Traditionally, C Band satellites ranged in power from 5～11 watts per transponder with receiving antenna of 5～12 feet in diameter. Satellite communication at C band is commonly used for voice communications, broadcast television, videoconferencing and radio.
2. **Baud (Bd)** is a kind of unit for signal transmitting speed, which means the number of total symbols transmitted by the communication system during a second. Thus, if a system transmits one symbol per second, it gets a signal transmission velocity of 1 Baud.
3. ***Streaming Media*** means the audio or video data transmitted as dataflow in a gradual and continuous manner via Internet, by means of small packets compress from signal data. In order to receive stream media efficiently, cable modem or DSL technique is required.
4. ***Primary Resource Clock*** **(*PRC*)** is a clock signal of very high precision and stability, which is distributed to clock of all levels through synchronous distribution network.
5. ***Frame Relay*** means a kind of technique of high speed network interface with frame as the least information transmitted unit, and transferring flow control, error correction etc. to intelligent terminal implemented. Similar to X.25, the use of variable length packet and multivariate statistics technique grants multiple terminals with Frame Relay technique the capability of sharing the common transmission media and bandwidth dynamically. Although it is somewhat difficult to guarantee the integrity of the data transmitted, frame relay technique is still capable of being relatively reliable in data transmission for many applications.
6. ***Service Network Interface*** **(*SNI*)** is a digital interface of wireless local loop system and exchange satisfying the access requirement of PSTN.
7. ***Service Control Point*** (SCP) is a term in Signaling System No. 7 (SS7). In order to offer quick and reliable services to customer, a SCP usually means a computer or an advanced exchange with a large-scale database.
8. ***Regeneration, Reshaping, Retiming*** **(*3R*)** means the basic three techniques for signal processing at receiving end. Regeneration ensures the output signal level of each connection enough to reach the next node. Reshaping is used for the elimination of pulse distortion caused by dispersion. Retiming can make the down clock recovery circuit receive signal exactly through reduction of time domain distortion.
9. ***Carrier to Noise Ratio*** **(*CNR*)** means the ratio of carrier signal level to noise level without any modulation. Some times, CNR is also expressed as C/N.
10. ***Plesiochronous Digital Hierarchy*** **(*PDH*)** is mainly designed for voice communication. Because it hasn't uniform standard of digital signal rate and frame structure internationally, it's

difficult for PDH to interconnect and intercommunicate. However, PDH equipments are universally deployed in spur tracks connected to the information superhighway.

11. ***Synchronous Digital Hierarchy*** (***SDH***) is a set of standardized standard digital signal structures for synchronous information transmitting, multiplexing, adding/dropping and cross connecting. Being proposed initially by U.S. Bellcore, SDH is designed to enable intercommunicating between equipments from various vendors, so that the network flexibility is improved. Being superior to PDH obviously, SDH serves as one of the fundamental techniques in the implementation of information superhighway.

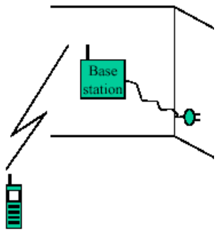
12. Being the integration of digital transmission network and digital switching network, ***Integrated Services Digital Network*** (***ISDN***) can carry various kinds of voice and data services. With the basic rate interface (BRI) including two independent B channels of 64Kbps respectively and a D channel of 16Kbps, you can get the rate of 64Kbps if you chose a B channel of ISDN to access Internet.

13. ***Broadband integrated services digital network*** (***B-ISDN***), developed from ISDN, can support various kinds of services with different transmitting speeds, including not only continuous services but also bursting services. The distribution scope of B-ISDN is very abroad, involving narrowband services with the rate upper limitation of 64Kbps, such as voice, FAX, broadband distributive services like broadcast, HDTV, broadband interactive communication services, i.e. Visual Phone, Video Conference, broadband bursting services, exp. High speed data, and so on.

14. ***Basic Rate Access*** (***BR***) is an interface rate of narrowband integrated service digital network (N-ISND) defined by ITU-T. Including two B channels, each of which is a bearing channel of 64Kbps, and a D channel as the digital command channel of 16Kbps, BR is also called 2B+D.

15. ***Electro Magnetic Compatibility*** (***EMC***) means the capability for equipment or system to operate normally in electromagnetic circumstance and generate EMI to other equipments in the circumstance under the extent of tolerance. Thus, EMC actually includes requirements from the following two contrary aspects.
    ① The EMI to surrounding circumstance generated by the equipment operating normally couldn't exceed a fixed limit;
    ② Equipment or system should have the capability of anti-EMI to a certain extent to EMI existing in its surround circumstance, which is also called ***Electro Magnetic Sensitivity*** (***EMS***).

16. ***Adaptive Differential Pulse Code Modulation*** (***ADPCM***) is an advanced PCM code method which can reduce the bit number of each analog signal sampling point from 8bits to 3 or 4bits to compress the signal, by calculating the difference between two consecutive speech samples from the voice signals. This calculation is encoded using an adaptive filter and as a result, is transmitted at a rate lower than the standard 64 Kbps technique. Typically, ADPCM allows an analog voice conversation to be carried within a 32Kbit digital channel. A recommended arithmetic by ITU-T in G.721 defines the compression method for 32 bit ADPCM with the sampling frequency of 8000Hz/s and 4bits for each sampling point. Compared with the conventional PCM code, this ADPCM method gets a pair gain of transmission capacity.

17. 

With wireless transceiver installed on the phone, ***Cordless Telephone*** usually constitutes a host phone and several associate phones. Because the similitude with the cordless associate phone and mobile phone set, this kind of telephone set is called by a joint name of cordless telephone (CT). Connected with PSTN directly, the host phone set owns functions like ordinary telephone. The associate telephones can be taken with person, receiving and transmitting calls within a certain region scope from the host phone through wireless connection with it. Thus, the associate phone of CT is also a kind of mobile phones functional for short distance.

18. ***Network Cheating*** is a network security protection method introducing mistaking resource to invaders and making them being ignorant of whether their aggressions success or not. At the same time, the protection system can also follow the invaders' action and remedy the possible security holes before being invaded.

19. ***Firewall*** is a kind of special network interconnection equipment protecting inner network circumstance and resource from being invaded lawlessly through more effective and serious control methods for visits between networks.

20. ***Content Filtering*** is an important function which can monitor information passing through the firewall and filter part of them according to the customer's requirement, such as garbage Emails, erotic information, reactive information or any other kind of information that the subscriber hopes to prohibit.

21. ***Virtual Private Dial-up Network*** (***VPDN***) means the virtual network implemented by dial-up method through public switched telephone network (PSTN), integrated service digital network (ISDN) and access network.

22. ***Virtual Private Network*** (***VPN***) means a kind of service offering customer all functions of private network through public telecommunication network. Subscribers do not have to set up the network or lease private line; even do not have to use private equipment, the private telecommunication network belonging to them can be built up.

23. ***Content Distribution*** is a kind of service aiming at those portal websites, which can make people in different places visit the nearest buffer server in order to reduce the load of host server and economize visitors' time of logging in.

24. ***Amplitude Shift Keying*** (***ASK***) is a kind of modulating method for binary signals, in which the carrier is shifted between Key-On and Key-Off in accordance with signal being modulated. So, Amplitude Shift Keying is also called ON-OFF Keying.

25. ***Bluetooth*** is a wireless LAN standard defined by its equipment manufacturers federally, working at 2.4GHz with the corresponding frequency scope of 10MHz and transmission rate of 1Mb/s.

26. ***Diversity Receiver*** (***DR***) is a measure to reduce the influence of fading by means of selecting and synthesizing outputs of more than two receivers with less cross-correlations. DR can be subdivided into space diversity, frequency diversity, time diversity etc.

27. ***Trellis Coded Modulation*** (***TCM***) is an advanced code-modulation method making sufficient

use of the redundancy generated from Convolutional code and memory effect of Viterbi Decoder. Thus, the signal sequence generated from level connected encoder and modulator has the longest free Euclidean Distance, which has the realistic decode method adopting Viterbi arithmetic.

28. ***Website*** is a fixed place in Internet for information issuance. Website constitutes domain name (namely IP address of the website) and the corresponding web-space, usually including home page and other pages with hyperlinks.

29. ***Network Management*** is an organ implementing the function of managing and controlling network. Through dynamic monitoring, organizing and controlling, Network Management can improve the demanded functions for better network service.

30. ***Hub*** works at the physical layer of the OSI reference model following the CSMA/CD visit method. The main function of Hub is to extend the transmitting distance through signal regenerating, reshaping and enlarging. In addition, Hub still has another function that collects all nodes surrounding a central node where it is.

31. ***DRAM*** is an abbreviation of **Dynamic Random-Access Memory**, the most widely used memory of computer. Because the DRAM can retain data in it for only a very little period, it's necessary for DRAM to refresh after a fixed time interval for fear that the data with it will be lost.

32. ***Extended Data Out*** **(*EDO*)** is another kind of memory which transmits data just after two clock cycles. Because it cancels the interval of two cycles between main-board and memory, EDO improves the corresponding velocity by 30% and reduces the time for data access to 60ns.

33. ***Synchronous Dynamic Random-Access Memory*** **(*SDRAM*)** is a kind of memory with the same controlling frequency of CPU, thus the RAM synchronizes the external frequency of CPU, causing the waiting time being canceled. So, the transporting velocity of SDRAM can achieve 7.5 ns, which is faster than that of EDO or DRAM.

34. ***Telecommunication Network*** means all the networks built and managed by the former Ministry of Posts and Telecommunications, such as conventional Public Switched Telephone Network (PSTN), Digital Data Network (DDN), Frame Relay (FR) network, Asynchronous Transfer Mode (ATM) network and so on.

35. ***Telecommunication supporting networks*** means all supporting networks with relevant monitoring and controlling signals transmission for the purpose of normal operation, functional enhancement and service quality improvement of telecommunication network, including synchronous network, common channel commands network, transmitting monitor network and so forth.

36. ***Interactive Service*** is a service offering bidirectional information exchange between subscribers or subscriber and host computer. Interactive Service can be subdivided into three service types: conversational service, message handling service and retrieval service.

37. ***Differentiated service*** **(*DiffServ*)** is a method for improving quality of service (QoS), which classifies the data flows of subscribers into different grades according to corresponding QoS

requirements. Thus, when it is time of network congestion, the data flow with higher grade has higher priority of queuing up and source impropriating.

38. ***Virtual Network Operator*** means a new kind of telecommunication operators which offers subscribers telecommunication services of reformed pattern or better quality. Without any telecommunication network resource, all kinds of services offered by Virtual Network Operator are complemented through the leased elementary equipments.

39. ***Universal Personal Telecommunications Number*** means the unique number which can distinguish a Universal Personal Telecommunication (UPT) subscriber and reach him. UPTN is not the physical number, which indicates the actual network address of customer terminal, but a logical number. Thus, with this UPTN, all the calls for you can reach you directly or not, in spite of wherever you being.

40. ***S-Video*** is a video interface of higher quality because of its avoidance of meaningless quality loss by means of canceling the signal overlapping. More concretely, S-Video processes the RGB tricolor signal and brightness signal respectively.

41. ***Personal Hand phone System*** is an extension of fixed telephone service. PHS is a wireless personal access system which can implement communication function depending on the primary fixed telephone switching without through mobile switching network.

42. ***Set Top Box*** is a kind of equipment for functional enhancement or expansion of TV. Because of usually being played on the top of TV set, it is called Set Top Box. STB can be subdivided into analog STB and digital STB, but now the STB always means the digital STB. With the conversion from digital signal to analog signal, STB can improve quality of TV programs and support almost all kinds of broadcast and interactive multimedia applications, such as watching ordinary TV programs, encrypted digital TV programs, ordering multimedia programs and information, electronic program guide, Email reception and transmission, Internet scanning, and network shopping etc.

43. ***Resolution*** means the displaying pixel number of a display, rested with the number of LCD points in display. Obviously, the denser the pixels are, the higher the resolution is, and the more distinct the showing picture is. For example, if a waiting picture of a mobile phone's resolution is 128×80, it implies that the mobile phone LCD has the maximum pixel of 128 at horizontal line and 80 at vertical line.

44. ***Next Generation Network*** **(*NGN*)** means the network which can support abundant multimedia information services, including voice, video, data etc. services, with open standard system architecture and the core technique of soft-switching. Making use of multiple broadband and QoS-enabled transport technologies, NGN offers its users with unrestricted access to various service providers, accompanied with additional support to generalization of mobile terminals. Consequently, NGN serves as a milestone in the history of telecommunication and symbolizes a brand-new era for telecommunication network.

45. ***Time Division-Synchronous Code Division Multiple Access*** **(*TD-SCDMA*)** is a kind of 3G mobile communication technologies with fully independent IPR owned by our country, having been adopted by ITU as one of the three 3G standards. Integrating various kinds of presently

advanced techniques, such as the intelligent antenna, synchronous CDMA, software Defined Radio (SDR) etc. TD-SCDMA has special superiorities in efficiency of spectrum utilizing, flexibility of business supporting, convenience of frequency allocating and system cost etc.

46. ***Computer Telecommunication Integration* (*CTI*)** technology has evolved from conventional Computer Telephony Integration (CTI) technology, with the present symbol "T" as "Telecommunication" in place of "Telephony". Thus, the present CTI technology not only offers conventional voice telephone service, but also supports other media types of information services, such as FAX, Email and so forth. The most representative services offered by CTI are the CPE information system, interactive voice response, call center system, value added service, IP phone and so on.

47. ***Information Communication Technology* (*ICT*)** means a new concept and technical domain formed from the amalgamation of information technology and communication technology.

48. ***Multimedia Communication*** is an outcome of the amalgamation of communicating and computing technology, a new communication method that can offer various types of media information at the same time, such as voice, graphic, picture, data, text and so on, during one communication process.

49. ***Local Multipoint Distribution System* (*LMDS*)** adopts a new broadband wireless access method, working at millimeter waveband with a frequency scope of 20～40GHz. Thus, LMDS can offer its subscriber the maximum access velocity of 155Mbps with a very broad working bandwidth. Because of its high frequency multiplexing efficiency and great system capacity, LMDS can offer various services such as voice, data and image etc.

50. ***Wireless Local Loop* (*WLL*)** is a wireless access method designed for network access with features of safety, quick and flexible installation, little building investment and maintaining cost, and so on. WLL has several different working frequency of 1.8GHz, 800MHz, 450MHz, even and 150MHz, which is decided for use according to the specific local spectrum arrangement.

51. ***Multiple-Input Multiple-output* (*MIMO*)** is an advanced technique for wireless channel fading restraining making use of multi-antenna, so it is also called Multiple-Transmission Multiple-Reception Antenna (MTMRA) technique. Because of the multiple-antenna and multiple-channel in both transmitting end and receiving end of MIMO system, the channel capacity increases pro rata to the number of antenna without any increase of bandwidth and transmission power. Consequently, with its improved spectrum utilizing efficiency, transmission reliability and reduced error code ratio, MIMO has been recognized as an absolutely necessary technique to next generation communication system.

52. ***Orthogonal Frequency Division Multiplexing* (*OFDM*)** is a branch of Multi-Carrier Modulation (MCM) with its channel being subdivided into some sub-channels, which is orthogonal with each other. At the transmitting end, high speed serial input of data signals are firstly converted into parallel low speed sub-data-signals, and then will be transmitted into the corresponding orthogonal sub-channel after being modulated. At the receiving end, these signals will be distinguished by correlating demodulation, which can reduce the Inter Carrier

Interference between channels.

53. ***Ultra Wide Band*** (***UWB***) is the most advanced wireless communication technology with the bandwidth and rate more than 1GHz and several hundreds Mbps respectively. The distinct feature of UWB is no-carrier, which extremely reduces the use of power for carrier transmission. Now UWB has been taken as a candidate of primary technology for Personal Area Network (PAN) by the IEEE802 Committee.
54. ***M-ary Quadrature Amplitude Modulation*** (***MQAM***) is a widely used modulation method in high capacity digital microwave communication system, which has very high spectrum utilizing ratio and relatively reasonable distribution of signal vector set while M is big enough. Because its relative facility for MQAM implemented, MQAM modulation such as 64QAM, 128QAM etc. have been widely used in high capacity digital microwave systems like SDH system, LDMS system and so forth.
55. ***Handoff*** means the all relative communication actions when a mobile station (MS) within a domain of a base station (BS) moves to domain of another BS. Handoff can be subdivided into hard handoff and soft handoff, the hard handoff causes the communication between MS and BS interrupted transitorily while the later ensures more than one BS keeping connection with the MS during the whole handoff period.
56. ***Instant Messaging*** is a kind of communication software with function of open interactive on-line communication tool making use of the Internet. There are many IM products now having become the integrated network communication tools, with the functional integration of E-mail box, mobile phone short message and file content services etc., which can offer subscribers real time text, voice and picture communication.
57. ***High Speed Downlink Packet Access*** (***HSDPA***) is a great technical advancement in history of 3G development, being brought forth for purpose of adaptation to the fast increasing High Speed Package Data Service and adopted within WCDMA. Using HSDPA method, the down-link data rate can be greatly improved from present 384 Kbps to 14Mbps, which is far more than the prescribed relevant rate of 2Mbps in 3G, with the same basic structure of WCDMA network. HSDPA is an unsymmetrical resolution approach which has been considered as the most hopeful technique among all kinds of wireless access approaches presently.
58. ***Enhanced Data Rate for GSM Evolution*** (***EDGE***) is an integration technology from GSM to 3G. Making use of new modulation method, EDGE can expand the signal space from 2 of present GSM modulated by GMSK to 8, and consequently make information contained in each symbol four times of that now.
59. ***Internet Protocol Television*** is an integration of various kinds of relevant techniques concerning Internet, multimedia, communication etc. Differing from the conventional analog Cable Antenna Television (CATV) and digital TV, IPTV is a new technology offering its customers various kinds of interactive information services including digital TV service.
60. ***Automatic Transfer Power Control*** (***ATPC***) makes the output power of transmitter automatically changing with the receiving voltage change at receiving end, in order to reduce interference to neighboring systems, upper attenuation, direct current power and error bit ratio,

and get the supererogatory output power increase of 2dB.

61．***Transmitter Diversity*** is an advanced technique that makes more than one Mobile Stations to obtain transmitting gains from one signal transmitted through two pairs of antenna. Because of supporting one point to multipoint transmission, Transmitter Diversity fits the developing requirement of mobile communication. The full name of this kind of technology is orthogonal transmitter diversity for reason of its use of simple orthogonal block coding method.

62．***General Packet Radio System* (*GPRS*)** is a kind of information service offering its mobile customers with the maximum data transmitting rate of 115Kbps. Customers of GPRS can keep state of on-line with the payment just according to the actual flux of information transmitted, without respect to the time of connection.

63．***Smart Antenna* (*SA*)** is a kind of antenna that is also called Intelligent Antenna, which offers wave bundles to target directly. For example, the mobile phone antenna is a kind of SA which can automatically adjust its effect factor, such as power etc. in accordance with the move of target.

Technical words and phrases

Metropolitan Area Network (MAN)　城域网
amalgamation　[əmælgə'meiʃən]　*n.* 融合，合并
Broadband integrated services digital network (B-ISDN)　宽带综合业务数字网
bursting service　突发业务
high definition television (HDTV)　高清电视
interactive communication　交互式通信，互通
Visual Phone　可视电话
Video Conference　电视会议
Frame Relay Line　帧中继线
local telephone exchange　本地电话交换机
C Band　C 波段
Carrier to Noise Ratio (CNR)　载噪比
Basic Rate Access (BR)　基本速率接口
ITU-T　国际电信联盟，国际电联
bearing channel　承载信道
Plesiochronous Digital Hierarchy (PDH)　准同步数字系列
intercommunicate　[intəkə'mju:nikeit]　*v.*　互相联络，互通消息
Primary Resource Clock (PRC)　基准主时钟
regenerator　再生器
damnification　[dæmnifi'keiʃən]　*n.* 损伤，损害
reshape　['ri:'ʃeip]　*vt.* 改造，再成形，采用新方针
reshaping　[电]（波形）整形
retiming　[电] 重定时

Service Control Point (SCP)　业务控制点
Signaling System No. 7 (SS7)　7 号信令系统
Synchronous Digital Hierarchy (SDH)　同步数字体系
Service Network Interface (SNI)　业务节点接口
Synchronous Optical Network (SONET)　同步光网络
Wireless Local Loop (WLL)　无线本地环路
Computer Telecommunication Integration (CTI)　计算机电信集成
Computer Telephony Integration (CTI)　计算机电话集成
Information Communication Technology (ICT)　信息通信技术
Time Division-Synchronous Code Division Multiple Access (TD-SCDMA)　时分同步码分多址
intelligent antenna　智能天线
synchronous CDMA　同步码分多址
software Defined Radio (SDR)　软件无线电
Streaming Media　流媒体
Wavelength Division Multiplexing　波分复用
wavelength　波长
fiber　['faibə]　*n*.(=fibre)　光纤
Frame Relay　帧中继
Electromagnetic Interference　电磁干扰
conduction　[kən'dʌkʃən]　*n*.　传导；输送
Electro Magnetic Compatibility (EMC)　电磁兼容性
anti-EMI　抗电磁干扰
Electro Magnetic Sensitivity (EMS)　电磁敏感性
Multiple-Input Multiple-output (MIMO)　多入多出技术
restrain　[ri'strein]　*vt*.　抑制，制止
multi-antenna　多天线
Multiple-Transmission Multiple-Reception Antenna (MTMRA)　多发多收天线（MTMRA）技术
Orthogonal Frequency Division Multiplexing (OFDM)　正交频分复用
Multi-Carrier Modulation (MCM)　多载波调制
Ultra Wide Band (UWB)　超宽带无线技术
Personal Area Network (PAN)　个人网络，领域网络
CDMA 450　大灵通（通信）
Diversity Receiver (DR)　分集接收
Enhanced Data Rate for GSM Evolution (EDGE)　增强型数据速率
Adaptive Differential Pulse Code Modulation (ADPCM)　自适应差分脉冲编码调制
Automatic Transfer Power Control (ATPC)　自动发送功率控制
pair gain　倍增
Smart Antenna (SA)　智能天线

host phone　主话机
Cordless Telephone (CT)　无绳电话
Nordic Mobile Telephone (NMT450)　北欧移动电话（系统）
jumping-off　['dʒʌmpiŋ'ɔf] *n.* 边远地区，文明终结，世界尽头，起点
High Speed Downlink Packet Access (HSDPA)　高速下行分组接入
High Speed Package Data Service　高速分组数据业务
unsymmetrical　[ʌnsi'metrikl]　*adj.*　非对称的，不匀称的
Internet Protocol Television　IP 电视
Transmitter Diversity　发射分集
General Packet Radio System (GPRS)　通用分组无线业务
Instant Messaging　即时消息，即时通信工具
flux　[flʌks]　*n.*　涨潮，变迁，[物]流量，通量
Quadrature Amplitude Modulation (MQAM)　*M* 进制正交幅度调制
Website　WWW 的站点
domain name　域名
home page　主页
hyperlink　*n.*　[计]超链接
Content Delivery Network (CDN)　内容分发网
Network Interface Card (NIC)　网卡，网络接口卡
Network Interface Adapter (NIA)　网络适配器
Network Management　网络管理
Service Level Agreement　服务水平协议
Public Switched Telephone Network (PSTN)　公共交换电话网
Telecommunication Network　电信网
Ministry of Posts and Telecommunications　邮电部
Digital Data Network (DDN)　数字数据网
Frame Relay (FR) network　帧中继网
Asynchronous Transfer Mode (ATM) network　ATM（同步传输模式）网
Telecommunication supporting network　电信支撑网
Interactive Service　交互式业务
bidirectional　[baidi'rekʃənəl]　*adj.*　双向的
conversational service　会话型业务
message handling service　消息型业务
retrieval service　检索型业务
Public Data Transmission Service　公共数据传输业务
Packet Switching Service　分组交换业务
Frame Relay Service　帧中继业务
Leased Circuit Service　租用电路业务

Bearer Service　承载业务
Local telephone service　本地电话业务
area code　电话区域代码，区号
Long distance telephone service　长途电话业务
payphone　付费电话
Supplementary service　补充业务
elementary telecommunication service　基本电信业务
Value Added Service (VAS)　增值业务
Differentiated service (DiffServ)　差异服务
Virtual Network Operator　虚拟网络营运商
Universal Personal Telecommunications Number　通用个人电信号码
Universal Personal Telecommunication (UPT)　通用个人通信
Electronic Payment　电子支付
Network Cheating　网络欺骗
Network Paralysis　网络瘫痪
Hit Count　点击次数
statistical　[stə'tistikl]　*adj.*　统计的，统计学的
IP Network　IP 网
Network Attack　网络攻击
Security Management　安全管理
security level　安全级别
Security Association (SA)　安全关联
Internet Protocol Security (IPsec)　互联网安全协议
firewall　防火墙
PSTN Firewall　电话网防火墙
anti-virus　*n.*　[计]反病毒程序，防（计算机）病毒
Telephone Firewall　电话防火墙
False Address Attack　错误地址攻击
Proxy Attack　代理攻击
Amplitude Shift Keying (ASK)　幅移键控
Key-On (Off)　按键开启（关闭）
ON-OFF Keying　开断键控
transceiver　[træns'siːvə]　*n.*　无线电收发机，收发器
Bluetooth　蓝牙
space diversity　空间分集
frequency diversity　频率分集
time diversity　时间分集
S-Video　S 视频端子

Trellis Coded Modulation (TCM)　网格编码调制
Viterbi Decoder　维特比译码器
Euclidean Distance　欧氏距离
Viterbi arithmetic　维特比算法
Set Top Box　机顶盒
encrypted digital TV programs　加密数字电视节目
electronic program guide　电子节目指南
sporadic　[spə'rædik]　*adj.*　零星的；不定时发生的；时有时无的；偶发性的
Personal Hand phone System　个人无线接入系统，无线市话，小灵通
Cable Antenna Television (CATV)　有线电视，共用天线电视
RGB tricolor signal　RGB 三原色

15.2　Reading Materials

1. WAPI

ISO turned down the Chinese technology, called WAPI (WLAN Authentication and Privacy Infrastructure), in voting to adopt the IEEE 802.11i security specification that was developed by the Institute of Electrical and Electronics Engineers Inc., according to a member of the IEEE 802.11 Working Group who asked not to be named because of working group rules.

The proposal to adopt WAPI as a standard was defeated, with eight votes in favor of the proposal and 17 against, according to a statement released by ISO. The adoption of 802.11i was approved by 24 votes in favor and three against. Voting ended on March 7.

The Chinese government said it would continue to support WAPI and that the rejection by ISO would not affect its domestic use in China.

2. DN and IMSI

Directory Number (DN) is the terminal number in a mobile communication system that the host caller has to dial up when the subscriber acts as the called side. Each DN is a digit sequence including country code, mobile access code, HLR identification code and the mobile user's number, all of which form the 13 digits together. For example, the country code of China is 86 and can be omitted on dialing interiorly; the mobile access code adopts network number scheme, and 134-139 belongs to China Mobile Corp. Ltd.

On the other hand, the International Mobile Subscriber Identification Number (IMSI) is a 15 digits sequence including mobile country code, mobile network code and the subscriber's identification code. For example, China's mobile country code is 460, and China Unicom’s mobile network code is 03.

3. Experts meet to promote cybersecurity and fight cybercrime

Geneva, 8 October 2007 — Experts from around the world gathered in Geneva to lay the

foundation for a global response to the constantly evolving nature of cyber-threats and the increasing level of sophistication of cybercrimes.

"Confidence and security in using information and communication technologies (ICT) are fundamental in building an inclusive, secure and global information society," said Dr Hamadoun Touré, Secretary-General of the International Telecommunication Union. "The legal, technical and institutional challenges posed by cyber-threats and cybercrime are global and far-reaching, and can only be addressed through a coherent strategy taking into account the role of different stakeholders and existing initiatives, within a framework of international cooperation." Dr Touré explained that the ITU Global Cybersecurity Agenda provides such an international framework.

4. ITU announces first global set of standards for IPTV

Geneva, 18 December 2007 — The International Telecommunication Union announced the first set of global standards for Internet Protocol TV (IPTV) today. The standards were built with technical contributions from leading service providers and manufacturers from the information and communication technology (ICT) sector and cement ITU's role as the global leader in IPTV standards development.

IPTV is one of the most highly visible services to emerge as part of the development of next-generation networks (NGN). Indeed, it is seen as both the business case and principal driver for accelerating deployment of NGN.

The new standards were developed by the Focus Group on IPTV (FG IPTV) in ITU's Telecommunication Standardization Sector (ITU-T).

Malcolm Johnson, Director of ITU's Telecommunication Standardization Bureau said, "Standards are crucial for IPTV to reach its market potential and global audience. They are necessary in order to give service providers — whether traditional broadcasters, ISPs, cable operators or telecoms service providers — control over their platforms and their offerings. Standards here will encourage innovation, help mask the complexity of services, guarantee quality of service, ensure interoperability and, ultimately, help players remain competitive."

5. ITU Global Symposium for Regulators Forges New Broadband Vision

Yasmine Hammamet, 17 November 2005 — The International Telecommunication Union's 6th annual Global Symposium for Regulators (GSR), held in Yasmine Hammamet, Tunisia from 14-15 November 2005, gathered regulators, policy makers and service providers from 110 countries to develop a new regulatory framework to promote broadband deployment and access in developing countries.

The advent of broadband has dramatically altered the ICT playing field, creating new opportunities for an ever-growing spectrum of players. The GSR's new vision for enhanced broadband deployment, which encompasses reduced regulatory burdens, innovative incentives and coordinated efforts, is designed to rapidly unleash commercial broadband deployment opportunities.

"There is not a significant environment on the planet in which broadband internet does not make sense, given the political will to foster an enabling environment," said Hamadoun I. Touré, Director of ITU's Telecommunication Development Bureau, in his opening address to the symposium."However, the pace of broadband take-up largely hinges on the regulatory framework."

15.3 Exercises

1. Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）宽带综合业务数字网
（2）载噪比
（3）基本速率接口
（4）7 号信令系统
（5）3R
（6）同步数字体系
（7）同步光网络
（8）可视电话
（9）计算机电信集成
（10）信息通信技术
（11）波分复用
（12）帧中继
（13）电磁干扰
（14）电磁兼容性
（15）时分同步码分多址
（16）正交频分复用
（17）多载波调制
（18）超宽带无线技术
（19）多入多出技术
（20）无绳电话
（21）分集接收
（22）自适应脉冲编码调制
（23）自动发送功率控制
（24）IP 电视
（25）切换
（26）*M* 进制正交幅度调制
（27）通用分组无线业务
（28）公共电话交换网
（29）通用个人电信号码
（30）数字数据网
（31）异步传输模式
（32）主页
（33）网站
（34）内容分发网络
（35）补充业务
（36）网卡

（37）虚拟网络营运商
（38）网络管理
（39）网络瘫痪
（40）点击次数
（41）安全关联
（42）动态随机存储器
（43）承载业务
（44）服务等级协议
（45）会话型业务
（46）本地电话业务
（47）长途电话业务
（48）幅移键控
（49）蓝牙
（50）网格编码调制
（51）小灵通
（52）有线电视
（53）电子节目指南

2．Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）Next (   ) Network is usually briefly expressed as (   ), which means the network which can support abundant multimedia information services with open standard system architecture and the core technique of soft-switching.

（2）Time Division-Synchronous Code Division Multiple Access can be briefly expressed as (   ). It is an internally recognized standard of (   ) mobile communication technologies with fully independent IPR owned by (   ).

（3）CTI means the integration technology of Computer and (   ) , with the most representative services as the CPE information system, (   ) voice response, call center system, value (   ) service and so on.

（4）ICT means a new concept and technical domain formed from the amalgamation of (   ) technology and (   ) technology.

（5）Orthogonal Frequency Division Multiplexing, briefly expressed as (   ), is a kind of Multi-Carrier Modulation with its channel being subdivided into sub-channels being (   ) with each other. At the transmitting end, high speed serial input of data signals are firstly converted into (   ) low speed sub-data-signals, and then will be transmitted into the corresponding orthogonal sub-channel after being modulated. At the receiving end, these signals will be distinguished by correlating demodulation, which can reduce the Inter Carrier (   ) between channels.

（6）Because Baud means the number of total symbols transmitted by the communication system during a second, signal transmission velocity of a system which transmits 100 binary symbols

per second is ( ) bps. The signal transmission velocity of a system which transmits 100 4-ary symbols per second is ( ) bps.

(7) Internet Protocol Television is an ( ) of various kinds of relevant techniques concerning the Internet, multimedia, ( ) etc.

(8) GPRS is the abbreviation of ( ), which is a kind of information service offering its mobile customers with the maximum data transmitting rate of ( ) Kbps. Customer of GPRS can keep state of ( ) with the payment just according to the actual flux of information transmitted, without respect to the time of ( ).

(9) Instant Messaging is a communication software with function of open ( ) on-line communication tool making use of ( ).

3. Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

(1) ( ) Regenerator regenerates and relays received signal for the purpose of overc-oming the distortion during the transporting process.

(2) ( ) Multimedia Communication is a new communication method that can offer multimedia information at the same time, such as voice, graphic, picture, data, text and so on, during one communication process.

(3) ( ) Streaming Media is a new means of multimedia information transmission.

(4) ( ) C Band expresses the frequency band scope of 3.4～4.2GHz.

(5) ( ) Carrier to Noise Ratio (CNR) and SNR are two concepts of almost the same meaning.

(6) ( ) In fact, Plesiochronous Digital Hierarchy (PDH) and Synchronous Optical Network (SONET) means the same technique.

(7) ( ) The distinct feature of UWB is no-carrier, which extremely reduces the use of power for carrier transmission.

(8) ( ) Multiple-Input Multiple-output (MIMO) is also called Multiple-Transmiss-ion Multiple-Reception Antenna (MTMRA). Because of the multiple-antenna and multiple-channel in both transmitting end and receiving end, the MIMO system improves spectrum utilizing efficiency, transmission reliability and reduced error code ratio.

(9) ( ) Wireless Local Loop is a wireless access method with different frequency of 1.8GHz, 800MHz, 450MHz, even and 150MHz, which is decided for use according to the different selection of specific individual user.

(10) ( ) Cordless Telephone usually comprises a host phone and several associate phones.

(11) ( ) Smart Antenna offers wave bundles to target round about.

(12) ( ) High Speed Downlink Packet Access (HSDPA) is a great technical advancement in 3G for purpose of adaptation to the fast increasing High Speed Package Data Service and adopted within WCDMA.

(13) ( ) Automatic Transfer Power Control (ATPC) makes the output power of transmitter automatically changing with the receiving voltage change at receiving end, getting the supererogatory output power increase of 2dB.

(14) (   ) Making use of new modulation method, EDGE can expand the signal space from 2 to 4, making information contained in each symbol two times of that now.

(15) (   ) The main function of Hub is to extend the transmitting distance through signal regenerating, reshaping and enlarging.

(16) (   ) Among DRAM, EDO and SDRAM, the fastest one is Synchronous Dynamic Random-Access Memory, and Dynamic Random-Access Memory is the slowest one.

(17) (   ) Telecommunication supporting networks include synchronous network, common channel commands network, transmitting monitor network and so forth.

(18) (   )Virtual Private Network (VPN) offers customers all functions of private network through public telecommunication network with high cost.

(19) (   ) S-Video is a video interface of higher quality by means of processing the RGB tricolor signal and brightness signal together.

(20) (   ) Trellis Coded Modulation (TCM) is an advanced code-modulation method making sufficient use of the redundancy generated from Convolutional code and memory effect of Viterbi Decoder.

(21) (   ) Set Top Box can be subdivided into analog STB and digital STB, but now the STB always means the analog one.

15.4 课文参考译文 基本概念

1．C Band——C 波段指常用于无线陆地微波和卫星通信的电磁频段。陆地微波通信中使用的 C 波段频率范围是 4～6GHz。卫星将 C 波段分为上行通道，频率大约是 6GHz，以及下行通道，频率大约是 4GHz。通常，C 波段卫星转发器的功率范围在 5～11W，相应的接收天线直径为 5～12 英尺。C 波段卫星通信主要应用于语音通信、视频会议和广播电视。

2．Bd——Baud，波特，一种信号传输速度的单位。它等于通信系统中每秒钟传送的码元数，所谓码元就是传送的一个符号，每秒钟传送一个符号的速率就是 1 波特。

3．Streaming Media，流媒体，指采用数据包的形式，在互联网上以数据流或连续方式传播的音频和视频信号数据。对于流媒体的有效接收，需要使用电缆调制解调或 DSL 等技术。

4．PRC——Primary Resource Clock，基准主时钟，高精度、高稳度时钟，该时钟经同步分配网分配给下面的各级时钟。

5．FR——Frame Relay, 帧中继, 是一种以帧为单位进行信息传输，并将流量控制、纠错等功能交由智能终端设备处理的高速网络接口技术，与 X.25 类似。由于采用可变长数据包技术和统计多元技术，帧中继可以使多个终端动态共享网络介质和带宽，虽然不能确保所传输数据的完整性，帧中继技术在实际应用中仍然具有较高的数据传输可靠性。

6．SNI——Service Network Interface，业务节点接口，是无线本地环路系统与交换机之间的接口，它是一数字接口，满足 PSTN 的接入需要。

7．SCP——Service Control Point，业务控制点，用于 SS7 的术语，为了提供快速可靠的服务，一个 SCP 通常是指包含大型数据库的一台计算机或先进的交换机。

8．3R——Regeneration, Reshaping ,Retiming，再生、重整形和重定时。再生保证每条连接的输出功率电平足以到达下一个节点。重整形消除色散等因素产生的脉冲失真。重定时消

除数字脉冲的时域失真，以使下行时钟恢复电路能准确地接收信号。

9．CNR——Carrier to Noise Ratio，载噪比，在没有经过任何调制之前，载波电平与噪声电平之比。也称为 C/N。

10．PDH——Plesiochronous Digital Hierarchy，准同步数字体系，主要是为语音通信设计，没有世界性统一的标准数字信号速率和帧结构，国际互联互通困难，已逐渐被同步数字传输体系 SDH 取代。但是在与信息高速公路相连接的支路和岔路上，PDH 设备仍被大量采用。

11．SDH——Synchronous Digital Hierarchy，同步数字传输体系，是一套可进行同步信息传输、复用、分插和交叉连接的标准化数字信号结构等级。SDH 最早由美国贝尔通信研究所提出，其目的是为了使不同厂家的产品能在光通路上互通，从而提高网络的灵活性。由于具有明显优于 PDH 的特点，SDH 技术成为实现信息高速公路的基础技术之一。

12．ISDN——Integrated Services Digital Network，综合业务数字网，由数字传输网和数字交换网综合而成，能用同一个网络承载各种语音和数据业务。ISDN 基本速率接口（BRI）包括两个独立的 64Kbit 的 B 信道和一个 16Kbit 的 D 信道，选择 ISDN 的 2B+D 端口的一个 B 信道上网，速度可达 64Kbps。

13．B-ISDN——broadband integrated services digital network，宽带综合业务数字网，是在 ISDN 的基础上发展起来的，可以支持各种不同类型、不同速率的业务，不但包括连续型业务，还应包括突发型宽带业务，其业务分布范围极为广泛，包括速率不大于 64Kbps 的窄带业务（如语音、传真），宽带分配型业务（广播电视、高清晰度电视），宽带交互型通信业务（可视电话、会议电视），宽带突发型业务（高速数据）等。

14．BR——Basic Rate Access，基本速率接入，ITU-T 定义为窄带 ISDN 的一种接口速率，也称为 2B+D，B 信道 64Kbit 为承载信道，D 信道 16Kbit 为数字信令信道。

15．EMC——Electro Magnetic Compatibility，电磁兼容性，指设备或系统在其运行电磁环境中符合要求运行并且不对其环境中的任何其他设备产生无法忍受的电磁干扰的能力。因此，一个设备的 EMC 实际上包括如下两个方面的要求：

① 设备在正常运行过程中对所在环境产生的电磁干扰（EMI）不能超过一定的限值;

② 设备对所在环境中存在的电磁干扰具有一定程度的抗扰力，即电磁敏感性（EMS）。

16．ADPCM——Adaptive Differential Pulse Code Modulation，自适应差分脉冲编码调制，一种较 PCM 先进的编码技术，通过计算编码语音信号中两个连续语音取样值之间的差异进行编码，从而将模拟采样的比特数从 8 位降低到 3～4 位。这种计算采用自适应滤波器进行编码，从而以低于标准 64Kbps 技术的速率进行传输。ITU-T 推荐 G.721 为 32 位 ADPCM 定义了一种算法（每秒 8000 次采样，每次采样为 4bit），与传统 PCM 编码相比，ADPCM 的传输容量加倍。

17．CT——Cordless Telephone, 无绳电话, 是一种在电话机内装有无线电收发机的电话，通常由一部主机和一部或几部副机组成。由于副机上没有电话机线，与移动电话手机相似，所以叫做无绳电话。主机与公用电话网相连的，内部装有无线收发机，具有普通电话机的功能，它与副机之间通过无线电波连接。副机可以随身携带，只要是在距离主机一定范围内，在任何地点都可以接受和呼叫，因此也是一种短距离的移动电话。

18．Network Cheating，网络欺骗是一种网络安全防护的方法，将入侵者引向一些错误的资源，使入侵者不知道其进攻是否奏效或成功，同时可跟踪入侵者的行为，在入侵者之前

修补系统可能存在的安全漏洞。

19．Firewall，防火墙，用来加强网络之间访问控制、防止外部网络用户非法进入网络、保护内部网络操作环境和资源的特殊网络互联设备。

20．Content Filtering，内容过滤。对经过防火墙的信息进行监视，并按照用户需求，滤除垃圾、带有色情、反动或任何用户希望禁止的信息。

21．VPDN——Virtual Private Dial-up Network，拨号虚拟专用网，利用拨号方式, 通过公用电话网（PSTN）或综合业务数字网（ISDN）和接入网实现的虚拟专用网。

22．VPN——Virtual Private Network，虚拟专用网，是指利用公用电信网络为用户提供专用网的所有各种功能。VPN 用户既不需要建设或租用专线，也不需要装备专用的设备，就能组成一个属于用户自己专用的电信网络。

23．Content Distribution，内容分发。针对各类门户网站而提供的因特网服务，使各地用户在访问这些网站时，可以访问最接近本地的缓存服务器，以节省时间和减轻网站服务器的负载。

24．ASK——Amplitude Shift Keying，振幅键控，一种键控技术，对应二进制调制信号，承载信号在开启和关闭之间切换，也就是常说的 ON-OFF 键控。

25．Bluetooth 蓝牙，一种由设备制造商联合制定的频率覆盖范围 10Mbit、工作频段为 2.4G、传输速率约 1Mbit 的无线局域网标准。

26．DR——Diversity Receiver，分集接收，将相关性较小的两路以上的收信机输出进行选择、合成，来减轻由衰落所造成的影响的一种措施。具体又可以分为空间分集、频率分集、时间分集等不同方式。

27．TCM——Trellis Coded Modulation，网格编码调制，一种高级的编码调制方法，它充分利用卷积编码中所产生的冗余度和维特比解码的记忆效应，使编码器和调制器级联后产生的编码信号序列具有最大的欧氏自由距离，而它的理想解码方式应采用维特比算法实现。

28．Website 网站，因特网上一块固定的面向全世界发布信息的地方，由域名（也就是网站地址）和网站空间构成，通常包括主页和其他具有超链接文件的页面。

29．Network Management 网管中心，执行网络管理和控制任务的机构，对网络资源进行动态监督、组织和控制，以提高网络良好服务所需的功能。

30．Hub，集线器，工作于开放系统互连 OSI 参考模型的第一层（物理层），采用 CSMA/CD 访问方式。主要对接收到的信号进行再生、整形和放大，以扩大网络传输距离，同时把所有节点集中在以它为中心的节点上。

31．DRAM（Dynamic Random-Access Memory），即动态随机存储器最为常见的系统内存。DRAM 只能将数据保持很短的时间。为了保持数据，DRAM 必须隔一段时间刷新一次。如果存储单元没有被刷新，数据就会丢失。

32．EDO——Extended Data Out，扩展数据输出，它取消了主板与内存两个存储周期之间的时间间隔，每隔两个时钟脉冲周期传输一次数据，大大地缩短了存取时间，使存取速度提高 30%，达到 60ns。

33．SDRAM——Synchronous Dynamic Random-Access Memory, 同步动态随机存储器，将 RAM 与 CPU 以相同时钟频率控制，使 RAM 与 CPU 外频同步，取消等待时间。所以其传输速率比 EDO 和 DRAM 更快, 高达 7.5ns。

34．Telecommunication Network——电信网是指原邮电部建设、管理的网，如传统的电

话交换网（PSTN）、数字数据网（DDN）、帧中继网（FR）、ATM 网等。

35．Telecommunication supporting networks——电信支撑网，保障电信网正常运行、增强网络功能、提高网络服务质量的支撑网路。支撑网中传递相应的监测和控制信号，包括同步网、公共信道信令网、传输监控网等。

36．Interactive Service 交互式业务，在用户间或用户和主机之间提供双向信息交换的业务，分为会话型业务、消息型业务和检索型业务三种。

37．DiffServ——Differentiated service，差异服务，一种改善服务质量的方法。它将用户的数据流按照服务质量要求来划分等级，当网络出现拥塞时，级别高的数据流在排队和占用资源时有更高的优先权。

38．VNO——Virtual Network Operator，虚拟电信运营商，本身没有电信网络资源，通过租用电信运营商的基础设施，以及对电信业务的改造加工，为用户提供电信业务的新型电信运营商。

39．UPTN——Universal Personal Telecommunications Number，通用个人电信号码。指唯一识别一个通用个人通信（UPT）用户并用来到达该用户的号码。UPTN 是一个逻辑号码而非标志用户终端实际网路地址的物理号码，即不论用户身处何处，使用这个号码，呼叫就可以接到或者转接达到此用户。

40．S-Video——一种信号质量更高的视频接口，它取消了信号叠加的方法，可有效避免一些无谓的质量损失。它的功能是将 RGB 三原色和亮度进行分离处理。

41．PHS——Personal Hand phone System，无线市话，俗称小灵通，是固定电话业务的延伸。PHS 是一种个人无线接入系统，无须经过移动交换网，只靠原有的固定电话交换机交换即可实现通信任务。

42．STB——Set Top Box，机顶盒，指用来增强或扩展电视机功能的一种设备，由于人们通常将它放在电视机的上面，所以被称为机顶盒。机顶盒有模拟机顶盒和数字机顶盒之分，现在一般所说的机顶盒都指数字机顶盒。数字电视机顶盒将数字电视信号转换成模拟信号以提供更高质量的电视节目，并支持几乎所有的广播和交互式多媒体应用，如收看普通电视节目、数字加密电视节目、点播多媒体节目和信息、电子节目指南（EPG）、收发电子邮件、因特网浏览、网上购物等。

43．Resolution, 分辨率，指 LCD 显示器所能表示的像素个数。像素越密，分辨率越高，图像越清晰。液晶显示器的分辨率取决于显示器中液晶点数量。例如，某手机待机图片分辨率为 128×80dpi，就表示手机显示器每行水平线上最多能表现 128 个像素，垂直线上最多有 80 行。

44．NGN——Next Generation Network，下一代网络，采用开放、标准的网络体系结构，以软交换技术为核心，采用开放、标准的体系结构，能够提供语音、视频、数据等多媒体综合业务的下一代网络。利用多种宽带传输能力和具有 QoS 保证的信息传送技术，NGN 可以使其用户自由地连接到不同的业务提供商。此外，NGN 还支持移动终端的通用性。可以说，NGN 是人类电信史上的一块里程碑，标志着一个全新的电信网络时代。

45．TD-SCDMA——Time Division-Synchronous Code Division Multiple Access, 时分同步的码分多址技术，是我国具有自主知识产权的第三代移动通信技术，也是被国际电联采纳的三个 3G 移动通信标准之一。该标准将智能天线、同步 CDMA 和软件无线电等当今国际领先技术融于其中，在频谱利用率、对业务支持的灵活性、频率配置的便利性及系统成本等方面

具有独特优势。

46．CTI——Computer Telecommunication Integration，计算机电信集成技术，由传统的计算机电话集成（Computer Telephony Integration）技术发展而来。现在，CTI 技术中的“T”指的是“Telecommunication”，这意味着 CTI 技术不仅要处理传统的电话语音，而且要处理包括传真、电子邮件等其他媒体形式的信息。目前， CTI 技术提供的典型业务主要有基于用户设备（CPE）的消息系统、交互语音应答、呼叫中心系统、增值业务、IP 电话等。

47．ICT——Information Communication Technology，信息通信技术, 是信息技术与通信技术相融合而形成的一个新概念和新技术领域。

48．Multimedia Communication——多媒体通信，指在一次呼叫过程中能同时提供多种媒体信息如声音、图像、图形、数据、文本等的新型通信方式。它是通信技术和计算机技术相结合的产物。

49．LMDS——Local Multipoint Distribution System，本地多点分配系统或本地多点分配系统。这是一种新的宽带无线接入方式，一般工作于 20～40GHz 的毫米波波段，具有很宽的工作带宽，用户接入速率最高可达 155Mb/s。它的频率复用度高、系统容量大，可以同时提供语音、数据、图像等多种业务。

50．WLL——Wireless Local Loop，无线本地环路, 专为接入网设计的无线接入方式，采用的频率一般在 1.8GHz，800MHz，450MHz，甚至还有 150MHz，视地区的频谱安排而定。有安装速度快、安装灵活方便、建设投资省、维护费用低、安全等特点。

51．MIMO——Multiple-Input Multiple-output，多入多出，是利用多天线来抑制无线信道衰落的技术，又称多发多收天线（MTMRA）技术。MIMO 系统的发射端和接受端均采用多天线和多通道，在不增加带宽和天线发射功率的情况下，使信道容量与天线数量成正比地增加，提高了频谱利用率和信道传输的可靠性，降低误码率。目前，MIMO 技术已被认为是新一代通信系统必须采用的一项技术。

52．OFDM——Orthogonal Frequency Division Multiplexing，正交频分复用，实际上是多载波调制（MCM）的一种。OFDM 将信道分成若干正交子信道，将高速数据信号转换成并行的低速子数据流，调制到在每个子信道上进行传输。在接收端采用相关解调技术来区分，以减少子信道之间的相互干扰（ICI）。

53．UWB——Ultra Wide Band，是超宽带无线技术，是一种使用 1GHz 以上带宽的最先进的无线通信技术，其速度可达几百 Mbps 以上。UWB 的特点在于不使用载波，大大减少了耗电量。IEEE 802 委员会已将 UWB 作为 PAN（personal area network）的基础技术候选对象来探讨。

54．MQAM——M-ary Quadrature Amplitude Modulation *M* 进制正交幅度调制，是大容量数字微波通信系统中普遍使用的载波调制方式。这种方式具有很高的频谱利用率，在调制进制数 *M* 较高时，信号矢量集的分布也较合理，实现起来也较方便。目前在 SDH 数字微波、LMDS 等大容量数字微波通信系统中广泛使用的 64QAM、128QAM 等均属于这种调制方式。

55．Handoff 切换，从一个基站向另一个基站转移用户站通信之动作，可分为硬切换和软切换两种。硬切换的特点是，通信短暂中断。软切换的特点是，一个以上的基站同时与同一个用户站保持通信。

56．IM——Instant Messaging，即时消息，又称即时通信工具，是一种利用因特网的通信软件，目前已成为一种开放互通的在线通信工具。现在已有众多 IM 产品，将电子信箱、手机短信、文件服务等内容融合起来，成为一种综合的网络通信工具，使其用户可通过互联网进行实时的文本、语音和图像通信。

57．HSDPA——High Speed Downlink Packet Access, 高速下行分组接入，是 3G 发展进程中的一项重大的技术进步, 是为了适应高速分组数据业务快速增长的需要而提出的，已被写入了 WCDMA。HSDPA 技术可以在不改变 WCDMA 基本网络结构的情况下大幅提高下行链路的数据速率，从目前的 384 Kbps 提高到 14 Mbps，远远超过了 3G 原定的速率（2Mbps）。HSDPA 是一种非对称的解决方案，是目前各种无线接入技术中最被看好的一种。

58．EDGE——Enhanced Data Rate for GSM Evolution，增强型数据速率，是一种从 GSM 到 3G 的过渡技术，通过在 GSM 系统中采用新调制方法，将现有 GSM 网络采用的 GMSK 调制技术的信号空间从 2 扩展到 8，从而使每个符号所包含的信息是原来的 4 倍。

59．IPTV——Internet Protocol Television，交互式网络电视, 一种集互联网、多媒体、通信等多种技术于一体，向家庭用户提供包括数字电视在内的多种交互式信息服务的技术。IPTV 既不同于传统的模拟有线电视，也不同于数字电视。

60．ATPC——Automatic Transfer Power Control 自动发送功率控制，使微波发信机输出功率在控制范围内自动跟踪接收端接收电平的变化而变化，以减少对相邻系统的干扰、减少上衰减问题、减低直流功率消耗、改善误码特性，并使输出功率额外增加 2dB。

61．Transmitter Diversity——发射分集，一种利用两副天线发射，实现同一发射信号使多个移动台获得发射增益的技术，支持点对多点的发射，适合移动通信发展的需要。这种技术利用了简单的正交分组编码方法，因此叫做“正交发射分集”，简称“发射分集”。

62．GPRS——General Packet Radio System，通用分组无线业务，为移动用户提供高达 115Kbps 的数据传输速率。GPRS 用户可一直处于在线状态，其收费也基于实际的数据传输量而并非连接时间进行。

63．SA——Smart Antenna，智能天线，一种天线提供直接指向目标的波束，比如移动电话的天线，能够随目标移动自动调整功率等因素。

Unit 16　Automotive Electronics

16.1　Text

Automotive electronics first began with the need for better controls for the engine. In fact, the first electronic parts in automobiles were used to control various engine functions and were referred to as ECUs (Engine Control Units). However, as electronic controls began to be used for other automotive applications, the acronym ECU took on the more general meaning "Electronic control unit". Today, specific ECUs are generally referred to as modules [e.g. the engine control module (ECM) or the Transmission Control Module (TCM)]. A modern car may have up to 100 electronic control units and a commercial vehicle up to 40.

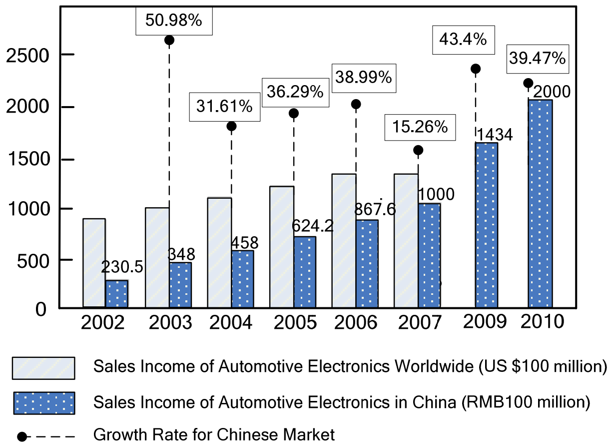


Figure 16-1　Growth of Automotive Electronics Market

16.1.1　Engine Electronics

An engine control unit (ECU), most commonly called the powertrain control module (PCM), is a type of electronic control unit that controls a series of actuators on an internal combustion engine to ensure the optimum running. It does this by reading values from a multitude of sensors within the engine bay, interpreting the data using multidimensional performance maps (called Look-up tables), and adjusting the engine actuators accordingly.

Before ECUs, air/fuel mixture, ignition timing, and idle speed were mechanically set and dynamically controlled by mechanical and pneumatic means. One of the earliest attempts to use such a unitized and automated device to manage multiple engine control functions simultaneously was created by BMW in 1939.

Modern ECUs use a microprocessor which can process the inputs from the engine sensors in real time. An electronic control unit contains the hardware and software (firmware). The hardware consists of electronic components on a printed circuit board (PCB), ceramic substrate or a thin laminate substrate. The main component on this circuit board is a microcontroller chip (CPU).

The software is stored in the microcontroller or other chips on the PCB, typically in EPROMs or flash memory so the CPU can be re-programmed by uploading updated code or replacing chips. This is also referred to as an (electronic) Engine Management System (EMS).

Sophisticated engine management systems receive inputs from other sources, and control other parts of the engine; for instance, some variable valve timing systems are electronically controlled. The Controller Area Network or CAN bus automotive network is often used to achieve communication between these devices.

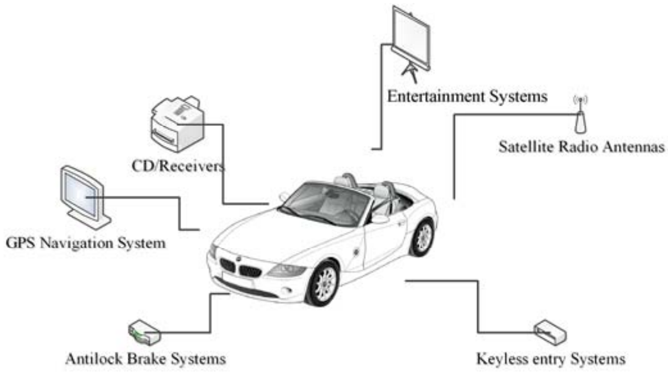


Figure 16-2   Automotive Electronics

Modern ECUs sometimes include features such as cruise control, transmission control, anti-skid brake control, and anti-theft control, etc. In 1988 Delco(GM's electronics division), had produced more than 28,000 ECUs per day, making it the world's largest producer of on-board digital control computers at the time.

16.1.2 Automotive Navigation System

An automotive navigation system is a satellite navigation system designed for use in automobiles. It typically uses a GPS navigation device to acquire position data to locate the user on a road in the unit's map database. Using the road database, the unit can give directions to other locations along roads also in its database. Dead reckoning using distance data from sensors attached to the drivetrain, a gyroscope and an accelerometer can be used for greater reliability, as GPS signal loss and/or multipath can occur due to urban canyons or tunnels. Some sorts can be taken out of the car and used hand-held while walking.

Automotive navigation systems were the subject of extensive experimentation, including some efforts to reach mass markets, prior to the availability of commercial GPS.

Most major technologies required for modern automobile navigation were already established when the microprocessor emerged in the 1970s to support their integration and enhancement by computer software. These technologies subsequently underwent extensive refinement, and a variety of system architectures had been explored by the time practical systems reached the market in the late 1980s. Among the other enhancements of the 1980s was the development of color displays for digital maps and of CD-ROMs for digital map storage.

16.1.3 Infotainment systems

In-Car Entertainment, (sometimes referred to as ICE, or IVI as in In-Vehicle Infotainment), is a collection of hardware devices installed into automobiles, or other forms of transportation, to provide audio and/or audio/visual entertainment, as well as automotive navigation systems. This includes playing media such as CDs, DVDs, TV and so on. Also increasingly common in ICE installs are the incorporation of video game consoles into the vehicle.

In-Car Entertainment systems have been featured in TV shows. In Car Entertainment has been become more widely available due to reduced costs of devices such as LCD screen/monitors, and the reducing cost to the consumer of the converging media playable technologies. Single hardware units are capable of playing CD, MP3, WMA, DVD.

16.1.4 Electric Vehicle

An electric vehicle (EV), also referred to as an electric drive vehicle, uses one or more electric motors or traction motors for propulsion. Three main types of electric vehicles exist, those that are directly powered from an external power station, those that are powered by stored electricity originally from an external power source, and those that are powered by an on-board electrical generator. Electric vehicles include electric cars, electric trains, electric lorries, electric motorcycles and so on.

Electric vehicles first came into existence in the mid-19th century, when electricity was among the preferred methods for motor vehicle propulsion, providing a level of comfort and ease of operation that could not be achieved by the gasoline cars of the time. The internal combustion engine (ICE) is the dominant propulsion method for motor vehicles but electric power has remained commonplace in other vehicle types, such as trains and smaller vehicles of all types.

During the last few decades, environmental impact of the petroleum-based transportation infrastructure, along with the peak oil, has led to renewed interest in an electric transportation infrastructure. Electric vehicles differ from fossil fuel-powered vehicles in that the electricity they consume can be generated from a wide range of sources, including fossil fuels, nuclear power, and renewable sources such as tidal power, solar power, and wind power or any combination of those.

Currently, though, there are more than 400 coal power plants in the U.S. alone, meaning that electric vehicle operation in regions serviced by these plants are effectively burning coal for locomotion. However it is generated, this energy is then transmitted to the vehicle through use of overhead lines, which necessarily involves transmission loss. The electricity may then be stored on board the vehicle using a battery. Vehicles making use of engines working on the principle of combustion can usually only derive their energy from a single or a few sources, usually non-renewable fossil fuels.

Technical words and phrases

engine control unit (ECU)　发动机控制模块

powertrain control module (PCM)　动力控制模块

actuators ['æktjueitə] *n.* 制动器
internal combustion engine 内燃机
look-up table 查找表
ignition timing [ɪg'nɪʃən 'taɪmɪŋ] 点火正时
idle speed ['aidl spi:d] 怠速
pneumatic [nu:'mætɪk,nju:-] *adj.* 充气的；气动的
unitized ['ju:nitaizd] *adj.* 组成的；合成的；成套的；成组的
printed circuit board (PCB) 印制电路板
ceramic substrate [si'ræmik 'sʌbˌstreit] *n.* 陶瓷基片
laminate ['læməˌneɪt] *n.* 层压材料；叠层，层压
Engine Management System ['endʒin 'mænidʒmənt 'sistəm] (EMS) 发动机管理系统
valve timing system 气门正时系统
Controller Area Network (CAN) 控制器局域网
bus [bʌs] *n.* 总线
anti-skid *adj.* 防滑的
brake [breik] *n.* 制动器，闸；刹车；
dead reckoning [ded 'rekəniŋ] *n.* 导航推测算法
drivetrain ['draivtrein] *n.* 动力传动系统
gyroscope ['dʒaiərəˌskəup] *n.* 陀螺仪
accelerometer [ækˌselə'rɔmitə] *n.* 加速器
fossil fuel ['fɔsl fjuəl] *n.* 化石燃料
tidal power ['taidl 'pauə] *n.* 潮汐能
locomotion [ˌləukə'məuʃən] *n.* 运动；移动；运动力；移动力
overhead line ['əuvəhed lain] 架空管道，架空线

16.2 Reading Material

1. China Automotive Electronics Industry in 2012

Following the rapidly growing demand for automobiles as well as the increasing requirements on automotive intelligence, Chinese automotive electronics market has witnessed robust development, with an AAGR of 29.5% in 2003—2011 and market size expected to outnumber RMB320 billion in 2012.

Concerning competitive landscape, foreign and joint-venture companies represented by Continental AG, Bosch, Denso and Delphi hold an absolutely dominant position, of which, Bosch Group enjoyed the biggest market share in 2011, and among the top ten corporations by share ranking, there was merely one local company called Shenzhen Hangsheng Electronics Co., Ltd., which was fixed on in-vehicle infotainment system products.

China Automotive Electronics Industry Report, 2012 by ResearchInChina selects two or three major kinds of automotive electronics products each from five categories, i.e. power control system, safety control system, body control system, ride control system and information system for research,

and makes analysis on market size, product demand, competitive landscape, development trends, etc..

In terms of the power control system market, engine management system (EMS) is the most important, with market size approximating RMB62.4 billion in 2011. Chinese EMS market is completely occupied by foreign corporations, among which, United Automotive Electronic System Co., Ltd (UAES) under Bosch Group seizes the largest share, followed by Delphi, Continental AG, Visteon, etc..

With respect to the safety control system, supplemental restraint system (SRS) and anti-lock braking system (ABS) are regarded as representative products of passive safety and active safety respectively. The quantity demand for SRS in China will reach 54.5 million units in 2012, while with the development of the automotive industry, ABS has basically been standard vehicle configuration in China, with matching proportion up to 91.01% in 2011.

(http://www.rnrmarketresearch.com)

2. Automotive electronics systems: trends and impact for test and measurement companies

When looking at electronic systems or, more specifically, ECUs (electronic control units) for these systems, however, growth is expected to be much stronger. Nowadays, electronics run pretty much everything in a vehicle. Between consumer love of electronic conveniences and hybrid or electric vehicles, the use of electronic systems in the automotive industry is accelerating at a furious pace. Of course, with new technologies come new challenges.

Among the design trends are faster transport buses; more wireless applications, often using a variety of standards; higher switching power, especially when talking about hybrid or electrical vehicles; and the sheer amount and density of electronics in modern vehicles, as shown in the example below.

Interoperability has emerged as a major challenge for automotive design engineers, and RF is now an integral part of that. Engineers need to look at not only analog and digital or serial data signals in the time domain but also, increasingly, RF signals in the frequency domain. Things like tracing the handshake between a radio transmitter and receiver as the communication is established and determining if a Bluetooth radio IC is transmitting when it is supposed to are common tasks that span the time and frequency domain. You now need to look at both domains at the same time. Time domain and frequency domain with time-correlated data are key to determining how one signal affects the other or if there are any unforeseen signal behaviors that might cause malfunctions.

（http://www.edn.com）

3. Early history of electric vehicles

Electric motive power started with a small drifter operated by a miniature electric motor, built by Thomas Davenport in 1835. In 1838, a Scotsman named Robert Davidson built an electric locomotive that attained a speed of four miles per hour (6 km/h). In England a patent was granted in 1840 for the use of rails as conductors of electric current, and similar American patents were issued to Lilley and Colten in 1847.

Between 1832 and 1839, Robert Anderson of Scotland invented the first crude electric carriage,

powered by non-rechargeable primary cells.

By the 20th century, electric cars and rail transport were commonplace, with commercial electric automobiles having the majority of the market. Over time their general-purpose commercial use reduced to specialist roles, as platform trucks, forklift trucks, ambulances,tow tractors and urban delivery vehicles, such as the iconic British milk float; for most of the 20th century, the UK was the world's largest user of electric road vehicles.

Electrified trains were used for coal transport, as the motors did not use precious oxygen in the mines. Switzerland's lack of natural fossil resources forced the rapid electrification of their rail network. One of the earliest rechargeable batteries - the nickel-iron battery - was favored by Edison for use in electric cars.

16.3 Exercises

1. Please translate the following Chinese words into English, and write out the corresponding English abbreviation if existing.

（1）发动机控制模块
（2）内燃机
（3）查找表
（4）怠速
（5）印制电路板
（6）发动机管理系统
（7）控制器局域网
（8）动力传动系统
（9）陀螺仪
（10）车载信息娱乐系统
（11）多径
（12）传输损耗

2. Read the following sentences carefully, and fill the brackets with correct words, phrases, abbreviations and numbers according to the text.

（1）The first electronic parts in automobiles were used to control various engine functions and were referred to as (　　　). Today, specific ECUs are generally referred to as the (　　　) control module or the (　　　) Control Module.

（2）ECU most commonly called the (　　　) control module (PCM), is a type of electronic control unit that controls a series of (　　　) on an internal (　　　) engine to ensure the optimum running.

（3）Before ECUs,(　　　), (　　　), and (　　　) were mechanically set and dynamically controlled by mechanical and pneumatic means.

（4）Modern ECUs use a (　　　) which can process the inputs from the engine (　　　) in real time. An electronic control unit contains the (　　　) and (　　　).

（5）The process of CPU being re-programmed by uploading updated code or replacing chips is also referred to as an (　　　).

（6）Modern ECUs sometimes include features such as (　　　) control, (　　　) control, (　　　) control, and (　　　), etc.

（7）Dead reckoning using distance data from sensors attached to the (　　　), a (　　　) and an (　　　) can be used for greater reliability.

（8）An electric vehicle (EV), also referred to as an electric drive vehicle, uses one or more (　　　)motors or (　　　) motors for (　　　).

（9）Electric vehicles differ from (　　　) fuel-powered vehicles in that the electricity they consume can be generated from a wide range of sources.

（10）Electric vehicle operation in regions serviced by coal power plants are effectively burning coal for (　　　). However it is generated, this energy is then transmitted to the vehicle through use of (　　　) which necessarily involves transmission (　　　).

3．Judge each the following description correct or not, and write your answer in the bracket behind the sequence number respectively.

（1）(　　　) Automotive electronics first began with the need for better controls for the engine.

（2）(　　　) A modern car may have up to 100 electronic control units and a commercial vehicle up to 40.

（3）(　　　) ECU controls a series of actuators on an internal combustion engine to ensure the optimum running by reading values from a multitude of sensors within the Steering wheel.

（4）(　　　) BMW use a unitized and automated device to manage multiple engine control functions simultaneously in 1942.

（5）(　　　) The main component on PCB contained in Modern ECUs is a microcontr-oller chip (CPU).

（6）(　　　) The Controller Area Network or CAN bus automotive network is often used to achieve engine propulsion between these devices.

（7）(　　　) GPS signal may encounter loss and/or multipath in urban canyons or tunnels.

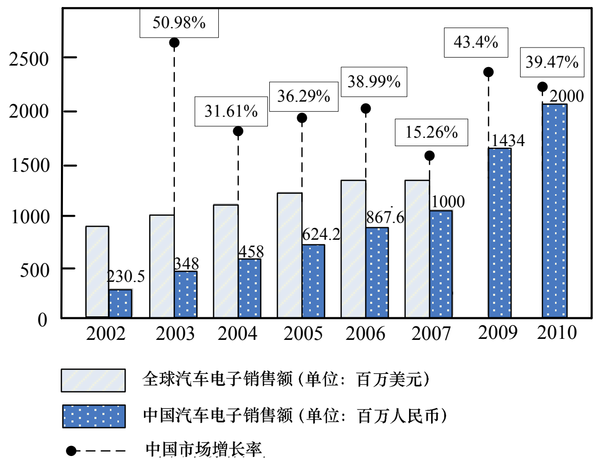
（8）(　　　) Most major technologies required for modern automobile navigation were already established when the microprocessor emerged in the 19th century.

（9）(　　　) There are six main types of electric vehicles.

（10）(　　　) Electric vehicles first came into existence in the mid-19th century.

16.4　参考译文　汽车电子技术

汽车电子技术最早用于汽车发动机控制，即早期汽车上的电子电路主要就是用于控制发动机的工作模式，因而被称为 ECU（发动机控制单元）。但随着汽车发动机以外的部件也开始启用电子控制后，ECU 的含义逐渐扩展到了“电子控制单元”。目前，常将 ECU 按模块功能进行划分，如引擎控制模块（ECM），传输控制模块（TCM）等。现代汽车最多有 100 个电子控制单元，而商用汽车中可达 40 个。



译图 16-1　汽车电子市场的增长情况

16.4.1　发动机电子设备

发动机控制单元（ECU）通常也称为动力控制模块（PCM），ECU 以电子的方式控制内燃机内部系列制动器以保持汽车的最佳行驶状态。通过读取汽车发动机内部多个传感器中的数据，调用多维查找表进行数据分析，从而对汽车发动机的制动器状态进行相应的调整。

在 ECU 出现之前，空燃比、点火正时和怠速是通过机械装置进行设定的，并以机械和气动的方式进行动态调整。德国宝马公司是最早尝试用全套自动设备实现发动机控制的汽车公司之一，该技术于 1939 年提出。

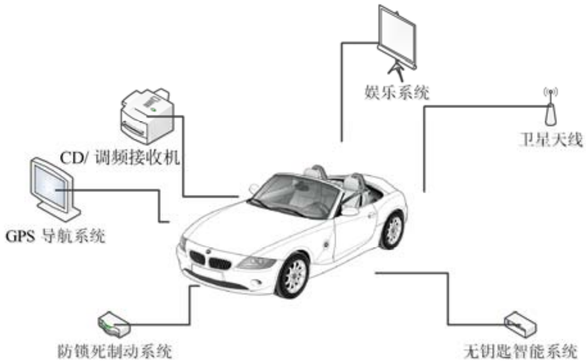
现代 ECU 由 CPU 来实现汽车发动机内部传感器数据的实时处理。ECU 包含硬件和软件（固件）两部分。硬件部分由印制电路板（PCB）为主，由陶瓷基板或薄层压板构成，PCB 上面焊接了各类电子元器件，电路板中最核心的电子器件是中央处理器（CPU）。软件程序存储在 PCB 的微控制器或其他芯片上，如 EPROM 或闪存中。这样，可以将更新后的程序代码下载到芯片或直接替换芯片，从而让 CPU 执行不同的程序功能，整个过程也称为发动机电子管理系统（EMS）。

高级发动机管理系统可从多个传感器源读取各类数据，从而控制发动机的各个环节。例如，某些可变气门正时系统即是电控设备，控制器局域网，即 CAN 总线就是用于实现汽车内部不同设备之间的通信。

当前市场上有些 ECU 甚至可以实现巡航、传输、制动防滑和防盗控制。1988 年，德科公司（通用汽车公司的下属子公司）每天的 ECU 产量达到了 28 000 个，是世界上最大的数字车载电脑生产商。

16.4.2　汽车导航系统

汽车导航系统基于卫星定位技术，即通过接收 GPS 卫星数据对行驶中的汽车进行定位，并将其显示于地图之中。通过调用道路数据库，导航系统可以给出如何达到目的地的具体路线。由于 GPS 信号在高楼林立的城市和隧道中传输将遭遇损耗和多径衰落，导航推测算法使用传动系统上传感器内部的距离信息，由陀螺仪和加速器进行修正以确保定位数据的精度。有些汽车导航仪还可以从汽车上取下来，让用户边走边用。



译图 16-2　汽车电子技术

汽车导航系统在商用之前需进行大量的实验测试，以满足大众市场的需求。

在 20 世纪 70 年代微处理器出现之后，汽车导航系统可以借助计算机软件提高其集成度和计算精度，因此，现代汽车导航仪所需的关键技术在那个时期就已经成熟。相关技术随后获得了极大的进展，各类系统如雨后春笋般出现，并在 20 世纪 70 年代后期进入市场。这一时期最重要的技术进步就是可以全彩色显示的数字地图，并可以使用 CD-ROM 存储地图数据。

16.4.3　车载信息娱乐系统

车载信息娱乐系统有时也简称为 ICE（或 IVI），是安装在汽车内部的一系列硬件设备，以提供音频、音视频娱乐服务以及汽车导航服务，如播放 CD/DVD、观看电视节目等。近年来，车载信息娱乐系统甚至可以玩电子游戏了。

车载信息娱乐系统的特色是可观看电视娱乐节目。由于 LCD 液晶屏成本以及相关媒体播放技术费用的下降，车载信息娱乐系统已逐渐大众化并普及开来。现在，仅单个设备就可以同时支持 CD、MP3、WMA 视频文件和 DVD 播放等功能。

16.4.4　电动汽车

电动汽车（EV），即用电力驱动的汽车，使用单个或多个电动机或牵引电机来驱动汽车。电动汽车主要有以下三大类：直接由外部电源供电驱动的电动汽车，由电池驱动的电动汽车和由车载发电机自行发电驱动的电动汽车，包括轿车、火车、卡车、摩托车等多种形式。

电动汽车最早出现于 19 世纪中期，那个年代曾热衷于使用电力来驱动汽车，与汽油发动的汽车相比，电力驱动的汽车在舒适度和操控性上更佳。目前，机动车主要使用内燃机驱动，但电力驱动在火车和其他一些小型机动车上仍有一席之地。

在过去的几十年里，由于消耗汽油、使用内燃机驱动的交通工具对环境造成的不良影响，以及石油产量的下降，使得人们逐渐重新关注起电动交通工具。与使用化石燃料驱动的汽车不同，电动汽车的发电方式较灵活，有些使用化石燃料、核能源发电，有些利用潮汐、太阳能或是风力等这些再生能源发电，甚至可以同时使用上述多种能源产生的电力。

目前，仅美国就有超过 400 家燃煤发电厂，这意味着这些发电厂服务区域内的电动汽车实质上是通过燃烧煤炭来获取电力，进而驱动汽车的。发电厂产生的电能通过管线传递至电动车。当然，这个传输过程会带来损耗，电能随后存储在汽车的蓄电池中。而使用内燃机驱动的汽车，其获取动力的方式较为单一，通常需要消耗不可再生的化石燃料。

附录 A　现代通信常用词组和缩写

2D-codes　二维码

800 service　被叫集中付费业务

3GPP (Third Generation Partnership Project)　第三代合作伙伴计划

A

access charge　接入费用

accessibility　可接入性

active network　主动网络

Ad hoc network　自组织网络，特定网络，对等网络

ADSL Termination Unit ADSL　终端设备

ADSL-Lite　简易经济型非对称数字用户线

adaptive modulation　自适应调制

additional POH　额外通道开销

affordability　可购性

Aloha　Aloha　协议

anti-interference　抗干扰

angle of incidence　入射角

anti-EMI　抗电磁干扰

application courier service　应用向导服务

application logic　应用逻辑

application middleware　应用中间件

application platforms　应用平台

application server　应用服务器

arbitrated loop　已裁定的环路

area code　电话区域代码，区号

availability　可获性

audio video information　音视频信息

AA——Adaptive Antenna,　自适应天线

ABD——Abbreviated Dialing　缩位拨号

ABF——Adaptive Beam Forming　自适应天线波束赋形技术

ABM——Asynchronous Balanced Mode　异步平衡方式

ABS——Alternate Billing Service　可选择记账业务

ABTS——Agreement on Basic Telecommunication Services　基础电信业务协议

ACD——Automatic Call Distribution　自动呼叫分配

ACK——ACKnowledge Character　确认字符

ACU——Automatic Calling Unit　自动呼叫装置

A/D——Analog / Digital conversion　模数转换
ADC——Analog to Digital Converter　模数变换器
ADM——Add/Drop Multiplexer　分插复用器
ADPCM——Adaptive Differential Pulse Code Modulation　自适应差分脉冲编码调制
ADSL——Asymmetric Digital Subscriber Line　非对称数字用户线
AF——Audio Frequency　音频、声频
AH——Application Hosting　应用托管
AM——Amplitude modulation　幅度调制，调幅
AM——Associated Memory　联想存储器
AMPS——Advanced Mobile Phone System　先进移动电话系统
AN——Access Network　接入网
AO/DI——Always On-line/Dynamic ISDN　永远在线/动态 ISDN
AOC——Advice of Charge　计费通知
AON——All-Optical Network　全光网络
AOWC——All-Optical Wavelength Converter　全光波长转换器
APD——Avalanche Photo Diode　雪崩光电二极管
APON——ATM Passive Optical Network ATM　无源光网络
ARQ——Automatic Repeat-reQuest　自动请求重发
AS——Anti-Spoofing　反电子欺骗
ASCII——American Standard Code for Information Interchange　美国信息交换标准码，ASCII 码
ASK——Amplitude Shift Keying　幅移键控
ASON——Automatic Switch Optical Network　自动交换光网络
ASP——Application Service Provider　应用服务提供商
ASP——Application Service Provider　应用服务提供者
ASTN——Automatic Switched Transport Network　自动交换传输网
ATM——Asynchronous Transfer Mode　异步传输模式
ATPC——Automatic Transfer Power Control,　自动发送功率控制
AVL——Automatic Vehicle Location,　自动车辆定位
AWG——American Wire Gauge　美国线规

B

backward (up) power control　反向功率控制
band-limited　频带受限的
bandwidth on demand　按需分配带宽
banner　网页标识
base-band　基带
bearing channel　承载信道
bending and scattering losses　弯曲和散射损耗
business-critical applications　企业要害应用

buffer　缓冲，缓冲器
bundled services　捆绑式服务
bursting service　突发业务
bus　总线，母线
BAS——Basic Assembler Program　宽带接入服务器
B-ISDN——Broadband Integrated Services Digital Network　宽带综合业务数字网
BR——Basic Rate Access　基本速率接口
BRAS——Broadband Remote Access Server　宽带远程接入服务器
BRI——Basic Rate Interface　基本速率接口
BSS——Broadcast Satellite Service　卫星广播业务
B-router——Bridge Router　桥接路由器

C

cable telephony　有线电视电话
cache　高速缓存器
call forwarding　呼叫前转
call screening　呼叫筛选
caller display　主叫显示
carrier synchronization　载波同步
C Band　C 波段
CDMA 450　大灵通（通信）
cell　信元
cellular mobile communication networks　蜂窝移动通信系统
Centrino　“迅驰”
channel coding　信道编码，纠错编码
checksum　校验和
chip　码片
chirp　啁啾
chromatic dispersion　色散，色度色散
circuit-switched　电路交换
circuit-switched networks　电路交换网
closed-loop　闭环
coherent-light　相干光
coherent reference carrier　相干基准载波
collect call　被叫方个人付费
conversational service　会话型业务
convolutional code　卷积码
Costas loop　科斯塔斯环
critical angle　临界角

cross-correlation　互相关性

cross-disturbance　交叉干扰

cyclic block code　循环码

codec　编译码器

concatenation　级联

CA——Certificate Authority　认证中心

CAC——Connection Admission Control　连接接纳控制

CAD——Computer Aided Design　计算机辅助设计

CAI——Computer Aided Instruction　计算机辅助教学

CAL——Computer Aided Learning　计算机辅助学习

CAMEL——Customized Applications for Mobile network Enhanced Logic　移动网定制应用增强逻辑服务器

CATV——Cable Antenna Television　有线电视，共用天线电视

CBE——Computer Based Education　以计算机为基础的教育

CBS——Cell Broadcast Service　小区广播业务

CBX——Computerized Branch Exchange　程控专用小交换机

CC——Call Center　呼叫中心

CCA——Common Communication Adapter　公用通信适配器

CCD——Charge Coupled Device　电荷耦合器件

CCS——Calling Card Service　电话卡业务

CCS——Common Channel Signalling　公共信道信令

CCS No.7——Common Channel Signalling No.7　7 号公共信道信令

CCSN——Common Channel Signalling Network　公共信道信令网

CCU——Communication Control Unit　通信控制器

CD——Compact Disk　光盘，激光唱盘

CDDI——Copper Distributed Data Interface　铜线分布式数据接口

CDMA——Code-Division Multiple Access　码分多址

CDN——Content Delivery Network　内容分发网

CELP——Code Excited Linear Predictive　线性码激励预测编码

CES——Circuit Emulation Service　电路仿真业务

CFD——Compact Floppy Disk　微型软磁盘

CGM——Computer Graphic Metafile　计算机图形元文件

CHILL——CCITT High Level Language CHILL　高级语言

CHTML——Compact HyperText Markup Language　压缩式超文本标识语言

CID——Calling Identity Delivery　主叫识别信息传送显示，来电显示

CIDR——Classless InterDomain Routing　无类别域际路由选择

CIF——Cells In Frames　帧中信元技术

CIR——Committed Information Rate　承诺信息速率，约定信息速率

CLI——Calling Line Identification　来电显示

CLIR——Calling Line Identification Restriction　主叫线路识别限制

CM——Cable Modem　电缆调制解调器

CMIP——Common Management Information Protocol　通用管理信息协议

CMIS——Common Management Information Service　通用管理信息服务

CMOS——Complementary Metal-Oxide Semiconductor　互补型金属氧化物半导体

CNAP——Calling Name Presentation　主叫名字显示（业务）

CNR——Carrier to Noise Ratio　载噪比

COBOL——Common business Oriented Language　COBOL 语言

CORBA——Common Object Request Broker Architecture　通用对象请求代理体系结构

COS——Card Operating System　卡片操作系统

COW——Cell On Wheels　车载基站

CP——Cordless Phone　无绳电话

CPN——Customer Premises Network　用户驻地网

CPU——Central Processing Unit　中央处理器

CRBTS——Customized Ring Back Tone Service　彩铃业务，炫铃业务

CRC——Cyclic Redundancy Check　循环冗余校验

CRM——Customer Relationship Management　客户关系管理

CRT——Cathode Ray Tube　阴极射线管

CS——Circuit Switching　电路交换

CSCW——Computer Supported Cooperative Work　计算机支持协同工作

CSMA——Carrier Sense Multiple Access　载波侦听多路访问

CSMA/CD——CSMA with Collision Detection　带冲突检测的 CSMA

CT——Cordless Telephone　无绳电话

CTI——Computer Telecommunication Integration　计算机电信集成

CTI——Computer Telephony Integration　计算机电话集成

Centrex——CENTRal EXchange　集中式用户交换机，虚拟用户交换机

CTU——Central Terminal Unit　中央终端设备

CUG——Closed User Group　闭合用户群

CWDM——Coarse Wavelength Division Multiplexing　稀疏波分复用

CoS——Class of Service　服务类别

cps——chip per second　码片速率单位，每秒码片

cps——cycle per second　频率单位，赫兹（每秒周数）

C-3、C3 system——Command、Control and Communication System　C-3 系统，指挥、控制和通信系统

D

differentially decode　差分编码

direct sequence spread spectrum　直接序列扩频调制

domain name　域名

data center　数据中心

data mart　数据商场

data mining　资料勘探

data mining tools　数据挖掘工具

data warehouse　数据仓储
datagram　数据报
DiffServ　区分服务
digital business　数字企业
digital certificate，also，digital signature　数字证书（也称数字签名）
digital decade　数字 10 年
domain name　域名
Doppler effect　多普勒效应
DSLAM　数字用户线路接入复用器
dB——Decibel　分贝
dpi——Dot per Inch　每英寸点数
D/A——Digital/Analog conversion　数模转换
DAA——Data Access Arrangement　数据接入装置
DACS——Data Acquisition and Control System　数据采集和控制系统
DAE——Data Acquisition Equipment　数据采集设备
DAF——Destination Address Field　目的地址字段
DAS——Direct Attach Storage　直接附加存储（技术）
DASD——Direct Access Storage Device　直接存取存储器
DBA——Dynamic Bandwidth Allocation　动态带宽分配
DBFA——Dual-Band Fiber Amplifier　双带光纤放大器
DBK——Data Base Key　数据库码
DBS——Direct Broadcasting Satellite　直播卫星（系统），直播卫星通信系统
DC——Data Communication　数据通信
DCC——Data Country Code　数据国家代码
DCE——Data Circuit-terminating Equipment　数据电路终端设备
DCF——Dispersion Compensating Fiber　色散补偿光纤
DCME——Digital Circuit Multiplication Equipment　数字电路倍增设备
DCS——Digital Cross-connect System　数字交叉连接系统
DD——Data Directory　数据字典
DDB——Distributed Data Base　分布式数据库
DDC——Direct Digital Control　直接数字控制
DDD——Direct Distance Dialing　长途直拨
DDE——Direct Data Entry　直接数据输入
DDN——Digital Data Network　数字数据网
DDP——Distributed Data Processing　分布式数据处理
DECT——Digital European Cordless Telecommunications　欧洲数字无绳电信系统
DEMUX——demultiplexer　解复用器，分路器
DES——Data Encryption Standard　数据加密标准
DFT——Discrete Fourier Transformation　离散型傅里叶变换
DIB——Directory Information Base　目录查询信息库，“电子查号”

DID——Direct Inward Dialing　直接拨入
DLC——Data Link Control　数据链路控制
DM——Data Multiplexer　数据多路复用器
DM——Delta Modulation　增量调制，Δ调制
DMA——Direct Memory Access　直接存储器存取
DMS——Discrete Memoryless Source　离散无记忆信源
DMS——Data Management System　数据管理系统
DMT——Discrete Multitone Modulation　离散多频音调制
DNA——Digital Network Architecture　数字网络体系
DNIC——Data Network Identification Code　数据网络识别码
DNS——Domain Name Service　域名服务
DOD——Direct Outward Dialing　直接向外拨号
DP——Data Packet　数据分组，数据包
DP——Data Processing　数据处理
DPCM——Differential Pulse Code Modulation　差值脉码调制
DPT——Dynamic Packet Transport　动态分组环技术
DQ——Directory Enquiry Service,　电子查号簿
DQDB——Distributed Queue Dual Bus　分布式队列双总线
DR——Diversity Receiver　分集接收
DRM——Digital Rights Management　数字权限管理
DS——Digital Signature　电子签名
DS——Data Stream　数据流
DSE——Data Switching Exchange　数据交换机
DSF——Dispersion Shifted Fiber　色散位移光纤
DSI——Digital Speech Interpolation　数字语声内插技术
DSL——Digital Subscriber Line　数字用户线（技术）
DSLAM——DSL Access Multiplexer　数字用户线接入复用器
DSP——Digit Signal Processing　数字信号处理（技术）
DSS——Double-Sideband Suppressed carrier Signal　抑制载波的双边带信号
DSS——Digital Signature Standard　数字签名标准
DSSS——Direct Sequence Spread Spectrum　直接序列扩频（技术）
DSU——Data Service Unit　数据服务单元
DTE——Data Terminal Equipment　数据终端设备
DTM——Dynamic synchronous Transfer Mode　动态同步传送模式
DTMF——Dual Tone Multi-Frequency　双音多频
DUN——Dial-Up Network　拨号上网（方式）
DVB——Digital Video Broadcast　数字视频广播
DVD——Digital Versatile Disc　数字（图像）光盘
DVI——Digital Video Interactive　数字视频交互技术
DWDM——Dense Wavelength Division Multiplexing　密集波分复用

DXC——Digital Cross Connection　数字交叉连接

E

electronic directory　电子号簿

electrical interference　电磁干扰

electrical isolation　电隔离，电气隔离

electronic program guide　电子节目指南

elementary telecommunication service　基本电信业务

emulation　仿真

emulator　仿真器

encoder　编码器

encrypted digital TV programs　加密数字电视节目

enhanced call routing　增强型呼叫选路

envelope detector　包络检波器

equalization　均衡

equalizer　均衡器

error-correction coding principle　纠错编码原理

Ethernet　以太网

excitation　激励

extended enterprise　外延的企业

Extranet　企业外联网

eye pattern　眼图

E-Marketplaces　电子市场

Erl——Erlang　爱尔兰（话务量单位）

EAI——Enterprise Application Integration　企业应用集成

EAROM——Electrically Alterable ROM　电可改写的只读存储器

EBCDIC——Extended Binary Coded Decimal Interchange Code　EBCDIC 码，扩充的二进制编码的十进制交换码

EBFA——Extended Band Fiber Amplifier　扩展带光纤放大器

EC——Electronic Business　电子商务

EC——Electronic Commerce　电子商务

ECB——Ethernet Client Bridge　以太网客户桥

ECSP——Electronic Commerce Service Provider　电子商务业务提供商

ED——Euclidean Distance　欧氏距离

EDFA——Er-Doped Fiber Amplifier　掺铒光纤放大器

EDGE——Enhanced Data Rates for GSM Evolution　增强型数据速率 GSM 演进技术

EDI——Electronic Data Interchange　电子数据交换

EDP——Electronic Data Processing　电子数据处理

EDPS——Electronic Data Processing System　电子数据处理系统

EFMA——Ethernet in the First Mile Alliance　以太网接入研究联盟

EFR——Enhanced Full Rate　增强全速率（技术）
EFTS——Electronic Fund Transfer System　电子资金转移系统
E-GPRS——Enhanced GPRS　增强型的 GPRS
EHF——Extreme High Frequency　极高频，毫米波
EI——Electromagnetic Interference　电磁干扰
EIRP——Equivalent Isotropic Radiated Power　有效全向辐射功率
E-LAN——Ethernet-based LAN　以太局域网
ELF——Extreme Low Frequency　极低频，极长波
EMC——Electro Magnetic Compatibility　电磁兼容性
EMI——Electro-Magnetic Interference　电磁干扰
EMP——Electro-Magnetic Pulse　电磁脉冲
EMS——Electro Magnetic Sensitivity　电磁敏感性
EMS——Enhanced Message Service　增值型消息业务
EMS——Electronic Mail System　电子函件系统
ENUM——Electronic NUMbering　电子号码
EOA——End Of Address　地址结束（符）
EOF——End Of File　文件结束（符）
EOM——End Of Message　报文结束（符）
EORPR——Ethernet Over Resilient Packet Ring　弹性分组环网上的以太网
EOS——Ethernet Over SDH/SONET　同步数字系列网络上的以太网
EOTD——Enhanced Observed Time Difference　高级时差检测定位技术
EP——Electronic Payment　电子支付
EPON——Ethernet Passive Optical Network　以太网无源光网络
EPROM——Erasable Programmable Read-Only Memory　可擦除、可编程只读存储器
EPS——Electronic Publishing System　电子出版系统
ERM——Enterprise Relationship Management　企业关系管理
ERP——Effective Radiation Power　有效发射功率
ERP——Enterprise Resource Planning　企业资源规划
ES——Earth Station　地球站
ESS——Electronic Switching System　电子交换系统
ETB——End of Transmission Block Character　信息块传送结束符
EVDSL——Ethernet VDSL　以太网超高速数字用户线

F

fast-hopped signal　快跳频（扩频）信号
fibre channel　光纤信道
fixed-weight code　等重码，恒重码
flow control　流量控制
forward power control　前向功率控制
frame　帧

Find me/Follow me　发现/跟踪
frame synchronization　帧同步
free space optics　自由空间光系统
frequency diversity　频率分集
frequency non-overlapping sub-channels　频率非重叠子信道
full-duplex communication　双工通信
FA——Frame Alignment　帧定位
FAA——False Address Attack　错误地址攻击
FAX——Facsimile　传真
FC——Flow Control　流控，数据流控制
FCFS——First Come First Service　先来先服务
FCSN——Fiber Channel Storage Network　光纤通道存储网络
FD——Full Duplex　全双工
FDMA——Frequency-Division Multiple Access　频分多址
FDD——Floppy Disk Drive　软磁盘机
FDDI——Fiber Distributed Data Interface　光纤分配式数据接口
FDM——Frequency Division Multiplexing　频分多路复用
FDMS——Frequency Division Multiple Access　频分多址
FEC——Forward Equivalence Class　转发等价类
FEC——Forward Error Correction　前向纠错
FEFO——First-Ended, First-Out　先结束、先送出
FET——Field Effect Transistor　场效应晶体管
FFT——Fast Fourier Transform　快速傅里叶变换
FH——Frequency Hopping　跳频
FHSS——Frequency Hopping Spread Spectrum　跳频扩频调制
FIFO——First In First Out　先进先出
FLC——Fixed Length Code　固定长度编码
FM——Frequency modulation　频率调制，调频
FMD——Follow Me Diversion　跟我转移
FOD——Fax On Demand　按需传真
FoIP——Fax over IP　IP 传真
FOMA——Freedom Of Mobile Multimedia Access　一种移动电话多媒体业务
FORTRAN——Formula Translator　FORTRAN 语言
FPH——Freephone Service　免费电话
FPLMTS——Future Public Land Mobile Telecommunication System　未来公用陆地移动通信系统
FR——Frame Relay　帧中继
FRL——Frame Relay Line　帧中继线
FSAN——Full Service Access Network　全业务接入网集团，全业务接入网
FSK——Frequency Shift Keying　频移键控，数字调频
FS——Fixed Service　固定业务

FSO——Free Space Optical communication　自由空间光通信

FSP——Full Screen Processing　全屏幕处理

FTP——File Transfer Protocol　文件传送协议

FTTB——Fiber To The Building　光纤到大楼

FTTC——Fiber To The Curb　光纤到路边

FTTH——Fiber To The Home　光纤到户

FW——Fire Wall　防火墙

FWA——Fixed Wireless Access　固定无线接入

FWM——Four Wave Mixing　四波混频

G

Gaussian white noise　高斯白噪声

geostationary orbit communication satellite　对地静止轨道

graded-index　梯度折射率

Gopher　菜单查询系统

GAP——General Assembly Program　通用汇编程序

GCSS——Global Communication Satellite System　全球通信卫星系统

GDSS——Group Decision Support System　群体决策支持系统

GEO——Geostationary Earth Orbit　地球同步轨道（卫星），静止卫星

GEOS——Gbit Ethernet Over SDH　SDH 网上的吉比特以太网（GbE）

GFP——Generic Framing Procedure　通用定帧法，通用成帧规程

GGG——Great Global Grid　网格

GGSN——Gateway GPRS Support Node　GPRS 支持节点网关

GII——Globe Information Infrastructure　全球信息基础设施，全球信息高速公路

GIS——Geography Information System　地理信息系统

GITH——Gigabit Internet To Home　吉比特因特网到家（网络）

GK——Gate Keeper　网闸，网守，网络管理器

GMPLS——Generalized Multi-Protocol Label Switching　通用多协议标签交换

GMSS——Geostationary Mobile Satellite Standard　地球同步移动卫星标准

GMT——Greenwich Mean Time　格林尼治标准时间

GOS——Grade Of Service　服务等级

GPON——Gigabit PON　吉比特无源光网络

GPU——Graphic Processing Unit　图形处理器

GPRS——General Packet Radio Service　通用分组无线业务

GPS——Global Position System　全球定位系统

GRE——Generic Route Encapsulation　通用路由封装（协议）

GSI——Grand Scale Integration　超大规模集成（电路）

GSM——Global System for Mobile communication　全球移动通信系统，“全球通”

GSMP——General Switch Management Protocol　通用交换机管理协议

GUI——graphical user interface　图形用户界面

GW——Gateway　网关，协议转换器

GbE,GE——Gigabit Ethernet　吉比特以太网

H

handshaking signal　握手信号

half-duplex communication　半双工通信

Hamming distance　汉明距

hard-switching　硬切换

hit count　点击次数

home page　主页

host　主机

host phone　主话机

host bus adapter　主机总线适配器

hosted outsourcing　托管给 ASP 的外包

hosting　托管

hub　集线器

Huffman source encoding algorithm　霍夫曼信源编码法

hybrid DS/FH　混合直接序列/跳频扩频

hypermedia　超媒体

hypertext　超文本

Home PNA——Home Phoneline Networking Alliance　1．家庭电话线网络联盟；2．家庭电话线组网技术

Home RF——Home Radio Frequency　家庭无线局域网技术

H-ARQ——Hybrid ARQ　混合 ARQ（协议）

HA——Home Agent　归属代理

HAN——Home Area Network　家域网

HCD——Home Country Direct　直拨对方国家话务员（业务）

HCI——Human Computer Interaction　人机交互作用

HCI——Human Computer Interface　人机界面

HDLC——High-level Data Link Control　高级数据链路控制（规程）

HDSL——High-bit-rate Digital subscriber Line　高比特率数字用户线

HDT——Host Digital Terminal　局用数字终端

HDTV——High Definition Television　高清电视

HDX——Half Duplex　半双工

HEOS——Highly Eccentric Orbit Satellite　高倾斜椭圆轨道卫星，椭圆轨道卫星

HF——High Frequency　高频，短波

HFC——Hybrid Fiber/Coax　混合光纤/同轴

HFT——Hand Free Telephone　免提电话

HIC——Hybrid Integrated Circuit　混合集成电路

HLR——Home Location Register　归属位置寄存器

HO——Hand Over　切换

HPC——Handheld Personal Computer　手持式个人计算机

HPCCT——High Performance Computing & Communication Program Initiative　高性能计算与通信计划

HRWG——Home RF Working Group　家用射频工作组

HSCSD——High Speed Circuit Switched Data　高速电路交换数据

HSDPA——High Speed Downlink Packet Access　高速下行分组接入

HSPDS——High Speed Package Data Service　高速分组数据业务

HTML——Hypertext Markup Language　超文本标识语言

HTTP——Hyper Text Transfer Protocol　超文本传送协议

I

IDSL ISDN　数字用户线路

IP Address　 IP 地址

IP Multicast　 IP 多播

IP PBX　基于 IP 的公司电话系统

IntServ　集成服务

intelligent device　智能设备

interactive　交互式的

interconnection　互连

interconnection charge　互连费

Internet　因特网，互联网

IP Switching　IP 交换技术

IP phone　IP 电话

information source coding　信源编码

i-Mode　I-Mode 业务

IP storage System　IP 存储系统

IA——Intelligent Antenna　智能天线

IAD—— Integrated Access Device　综合接入设备

IC—— Integrated Circuit　集成电路

IC card—— Integrated Circuit Card　IC 卡

ICC——Internet Call Center　因特网呼叫中心

IVDS——Interactive Video Data Service　交互式视像数据业务

ICMP——Internet Control Message Protocol　网间控制信息协议

ICP——Internet Content Provider　因特网内容提供商

ICS——Incoming Call Screening　来话筛选（业务）

ICT——Information Communication Technology　信息通信技术

ICW——Internet Call Waiting　因特网呼叫等待

IDC——Internet Data Center　因特网数据中心

IDN——Integrated Digital Network　综合数字网

IDSL——ISDN Digital Subscriber Line　综合业务数字网数字用户线，ISDN 数字用户线

IE——Internet Explorer　因特网浏览器

IEEE——Institute of Electrical and Electronics Engineers　美国电气和电子工程师协会
IEP——Internet Equipment Provider　因特网设备提供商
IETF——Internet Engineering Task Force　因特网工程任务组
IF——Intermediate Frequency　中频
IFS——International Freephone Service　国际被叫集中付费业务，国际“免费电话业务”
IM——Instant Messenger　即时送信业务
IMA——Inverse Multiplexing over ATM　IMA 技术，ATM 反向多路复用
IMEI——International Mobile Equipment Identity　国际移动设备认证（码）
IMSI——International Mobile Subscriber Identification Number　国际移动用户识别码
IM——Instant Messaging　即时消息，即时通信工具
IMT-2000——International Mobile Telecommunication-2000　国际第三代移动通信系统
IN——Intelligent Network　智能网
I/O——Input/Output　输入/输出
ION——Integrated On-demand Network　集成请求式网络，综合式按需服务网络
Internet of Things　(IoT)——物联网
ION Intelligent Optical Network　智能光网络
IP——Internet Protocol　网际协议，因特网协议
IPsec——Internet Protocol Security　互联网安全协议
IPTV——Internet Protocol Television　IP 电视
IP UMTS——All IP UMTS　全 IP 通用移动通信系统
IP VPN——IP Virtual Private Network　IP 虚拟专用网
IP ng——IP next generation　下一代 IP
IP sec——Internet security Protocol　因特网安全协议
IPN——Internet Personal Number　IPN 业务（因特网个人号码业务）
IPOA——IP Over ATM　ATM 网上的 IP 技术
IPOS——IP Over SDH　SDH 网上的 IP 技术
IPX/SPX——Internetwork Packet Exchange/Sequence Packet Exchange　互联网包交换/顺序包交换（协议）
IPv4——Internet Protocol version 4　网际协议版本 4
IPv6——Internet Protocol Version 6　网际协议版本 6
IRA——International Reference Alphabet　国际参考字母表
IRC——Internet Relay Chat　英特网聊天
ISC——International softswitch Consortium　国际软交换协议
ISDN——Integrated Services Digital Network　综合业务数字网
ISI——Inter-Symbol Interference　码间干扰
ISM——Industrial/Scientific/Medical　ISM 频段，工业/科学/医药（频段）
ISP——Internet Service Provider　因特网服务提供商
ISV——Independent Software Vendor　独立软件提供者
IT——Information Technology　信息技术
ITA——Information Technology Agreement　信息技术协议
ITV——Interactive CATV　交互式有线电视

IVD——Integrated Voice & Data　综合话音和数据
IVOD——Interactive Video On Demand　交互式视频点播
IVR——Interactive Voice Response　交互式语音应答
IrDA——Infrared Data Association　1.红外线数据标准协议，2.红外线点到点通信技术
iCRM——Internet CRM　利用因特网技术的 CRM
iNOW——Interoperability Now　iNOW（I 电话互联互通协议）
iSCSI——Internet Small Computer System Interface　互联网小型计算机系统接口

J

jitter　抖动
jumper　跳线
JET——Just Enough Time　“恰量时间”（协议）
JPEG——Joint Photographic Experts Group　联合图像专家组，静止图像压缩编码国际标准
JVT——Joint Video Team 联合视频组
Java——Java Language　Java 语言

K

Key-On　按键开启
Key-Off　按键关闭

L

label　标签，标记
leaving words　留言
local telephone exchange　本地电话交换机
local telephone service　本地电话业务
long distance telephone service　长途电话业务
lower sideband　下边带
LA——Location Area　位置区
LAN——Local Area Network　局域网
LAN-based IP network　局域 IP 网
LCS——Leased Circuit Service　租用电路业务
LPC——Liner Predictive Coding　线性预测编码
LPI——Low Probability of Intercept　低截获概率
LS——Location Service　定位服务
LSI——Large-Scale-Integrated circuit　大规模集成电路

M

matched filter　匹配滤波器
maximum DSL speeds　最高 DSL 速率

message handling service　消息型业务
mirror site　镜像站点
multi-access　多址接入
modal dispersion　模间色散，模式色散
multi-antenna　多天线
multimedia communication　多媒体通信
multi-mode graded-index fiber　多模梯度光纤
multi-mode step-index fiber　多模阶跃光纤
multi-user communication system　多用户通信系统
MAC——Media Access Control　媒体接入控制
MAHO——Mobile Assisted HandOff　移动台辅助切换
MCM——Multi-Carrier Modulation　多载波调制
MF——Mobile FM　移动调频
MIMO——Multiple-Input Multiple-output　多入多出技术
MMS——Multimedia Messaging Service　多媒体信息业务
MPT——Ministry of Posts and Telecommunications　邮电部
MPTY——Multi-party Telecommunication　多方通信
MS——Multimedia Service　多媒体业务
MSC/VLR——Visitor Location Registry of Mobile Switching Center　移动交换中心的访问位置寄存器
MTMRA——Multiple-Transmission Multiple-Reception Antenna　多发多收天线技术

N

noise rejection　噪声抑制，去噪，噪声剔除
Nyquist condition for zero ISI　奈奎斯特无码间干扰条件
No.7 command　7 号信令
No.7 network　7 号信令网
non-distortion coding principle　不失真编码原理
Nyquist's Theorem　奈奎斯特定理
NA——Network Attack　网络攻击
NAP——Network Access Point　网络接入点
NAS——Network-Attached Storage　直接挂网的存储器
NC——Network Cheating　网络欺骗
NC——Network Connection　网络连接
NEL——Network Element Layer　网元层
NIA——Network Interface Adapter　网络适配器
NIC——Network Interface Card　网卡，网络接口卡
NM——Network Management　网络管理
NMT450——Nordic Mobile Telephone　北欧移动电话（系统）

NP——Network Paralysis　网络瘫痪

O

off hook　摘机

off-line　掉线，断线

oneness　单一化

one-stop shopping　一站购齐

ON-OFF Keying　开断键控

open-loop　开环

optical fiber communications　光（纤）通信

outsourcing　外包

over-the-air　空中下载

ODSI——Optical Domain Service Interconnect　光域业务互连

OFDM——Orthogonal Frequency Division Multiplexing　正交频分复用

OVPN——Optical Virtual Private Network　光虚拟专用网

P

packet-switching　分组交换，包交换

packet-switched network　分组交换网

partial-response signal　部分响应信号

phase ambiguity　相位模糊度

phase-coherent　相位相干

physical connecting　物理连接

p-n junction diode　p-n 结二极管

pilot signal　导频信号

portal　入口、门户

power spectral density　功率谱密度

PSTN Firewall　电话网防火墙

pulse-averaging discriminator　脉冲平均鉴别器

PA——Proxy Attack　代理攻击

PABX——Private Automatic Branch exchange　自动用户小交换机

PAN——Personal Area Network　个人网络，领域网络

PBX——Private Branch Exchange　专用分局交换机

PCM——pulse code modulation　脉冲编码调制

PDA——Personal Digital Assistant　个人数字助理，掌上电脑

PDH——Plesiochronous Digital Hierarchy　准同步数字系列

PDTS——Public Data Transmission Service　公共数据传输业务

PHS——Personal Hand phone System　个人无线接入系统，无线市话，小灵通

PLL——Phase Locked Loop　锁相环

PM——Phase Modulation　调相，相位调制
PNS——Pseudo-random Sequence　伪随机序列
PPC——Peer-to-Peer Computing　对等计算
PRC——Primary Resource Clock　基准主时钟
PSS——Packet Switching Service　分组交换业务
PSTN——Public Switched Telephone Network　公共交换电话网
PTO——Public Telecommunication Operator　公众电信运营者

Q

QCELP——Qualcomm Code Excited Linear Prediction　码激励线性预测编码
QoS——Quality-of-Service　服务质量

R

real time voice service　实时语音业务
retrieval service　检索型业务
resource reservation protocol　资源预留协议
routing policy　选路策略
RADSL　速率自适应数字用户线路
RGB tricolor signal　RGB 三原色
RAID——Redundant Array of Inexpensive Disks　经济磁盘冗余阵列
RAM——Random Access Memory　随机存取内存, 随机存取存储器
Radio-frequency identification (RFID)——电子标签
ROM——Read Only Memory　只读存储器
RTP——Real-time Transport Protocol　实时传输协议

S

satellite communication　卫星通信
SDSL　对称数字用户线路
security level　安全级别
self-synchronization　自同步
set top box　机顶盒
Shannon Formula　香农公式
Shannon-Fano source encoding algorithm　香农-范诺信源编码法
signal-to-noise ratio （SNR）　信噪比
simplex communication　单工通信
single sideband（SSB）　单边带
slope filter　斜率滤波器
slow-hopped signal　慢跳频信号
soft switching　软切换

source encoding　信源编码

space diversity　空间分集

space communication　宇宙通信，空间通信

space station　宇宙空间站，太空站

step-index　阶跃折射率

streaming media　流媒体

sub-channel　子信道

successful connection ratio　成功接入率

S-Video　S 视频端子

symbol rate　码元速率

symbol synchronization　码元同步，位同步

system capacity　系统容量

synchronous CDMA　同步码分多址

SA——Smart Antenna　智能天线

SA——Security Association　安全关联

SAN——Storage-Area Network　存储域网

SBC——Sub Band Coder　子带编码器

SCP——Service Control Point　信令控制点

SCSI——Small Computer Systems Interface　小型计算机系统接口

SDH——Synchronous Digital Hierarchy　同步数字体系

SDR——Software Defined Radio　软件无线电

SDRAS——Satellite Digital Audio Radio Service　卫星数字音频无线电业务

SMSCB——SMS Cell Broadcast　短消息群发

SHR——Self-healing Hybrid Ring　SDH 自愈环

SLA——Service Level Agreement　服务水平协议

SM——Security Management　安全管理

SME——Short Message Entity　短消息实体

SMS——Short Messaging Service　短消息服务

SM-MO——Mobile Originated Short Message　由移动台发起的短消息

SM-MT——Mobile Terminated Short Message　到达移动台的短消息业务

SMR/PMR——Specialized/Private Mobile Radio Service　专用移动无限业务

SMS——Short Message Service　短消息业务

SMSC——Short Message Service Centre　短消息服务（业务）中心

SoC——System on Chip　片上系统

SONET——Synchronous Optical Network　同步光网络

SS——Storage Service　存储业务

SS——Supplementary service　补充业务

SS7——Signaling System No. 7　7 号信令系统

SSP——Service Switching Point　信令交换点
SSMA——Spread-Spectrum Multiple Access　扩频多址

T

talk-range　通话距离
Taylor series　泰勒级数
tele-service　用户终端业务
time diversity　时间分集
TACS——Total Access Communication System　全接入通信系统
TCM——Trellis Coded Modulation　网格编码调制
TCP——Terminal Connecting Points　终端连接点
TD——Transmitter Diversity　发射分集
TDMA——Time-Division Multiple Access　时分多址
TD-SCDMA——Time Division-Synchronous Code Division Multiple Access　时分同步码分多址
TN——Telecommunication Network　电信网
TS——Telecommunication Service　电信业务
TSN——Telecommunication Supporting Network　电信支撑网

U

undetected error　未被检验出的错误
underwater acoustic network　水下声讯系统
unit impulse　单位冲激
universal access　普遍接入
universal service　普遍服务
UltraDWDM　超密集波分复用
upper sideband　上边带
UADSL——Universal Asymmetrical Digital Subscriber Line　通用不对称数字用户线路
UIM——User Identity Model card　用户识别模块卡
UMTS——Universal Mobile Telecommunications System　通用移动通信系统
UPT——Universal Personal Telecommunication　通用个人通信
UPTN——Universal Personal Telecommunications Number　通用个人电信号码
UWB——Ultra Wide Band　超宽带无线技术

V

value added service　增值业务
videoconference　视频会议
virtual workplaces　虚拟工作场所
Viterbi arithmetic　维特比算法

Viterbi Decoder　维特比译码器
voice mailbox　语音信箱
VC——Video Conference　电视会议
VCO——Voltage-Controlled Oscillator　压控振荡器
VDSL——Very high data rate Digital Subscriber Line　超高速数字用户线路
VLF——Very Low Frequency Band　甚低频
VLIW—— very long instruction word　超长指令字
VLSI——Very-Large-Scale-integrated Circuit　超大规模集成电路
V-mail——Video mail　视频邮件
VoD——Video on Demand　视频点播
VoIP——Voice over Internet Protocol　IP 语音业务
VOT——Vote Over Telephony　电子(话)投票
VP——Video Phone　可视电话
VR——Virtual Reality　虚拟现实

W

web cast　网播
website　网站
webpage　网页
wireless communication　无线通信
WAP——Wireless Application Protocol　无线应用协议
WDM——Wavelength Division Multiplexing　波分复用
Wi-Fi——Wireless Fidelity　无线保真
WiMAX—— Worldwide Interoperability for Microwave Access　全球微波互联接入
WLL——Wireless Local Loop　无线本地环路
WPA——Wi-Fi Protected Access　Wi-Fi 网络安全存取

Z

zero-crossing detector　过零鉴别器

附录 B　常见国际电信组织机构

ACEC—advisory committee on electromagnetic compatibility　电磁兼容顾问委员会

ANSI—American National Standard Institute　美国国家标准局 U.8

ARIB—Association of Radio Industry Businesses　无线行业企业协会

ARPA—Advanced Research Projects Agency　（美）高级研究计划署

CCIR—International Radio Consultative Committee　国际无线电咨询委员会

CCITT—Consultative Committee, International Telegraph and Telephone　国际电报电话咨询委员会

China Mobile　中国移动

China Unicom　中国联通

China Telecom　中国电信

CISPR—international special committee on radio interference　国际无线电干扰特别委员会

CITT—International Telegraph and Telephone Consultative Committee　国际/电报咨询委员会

CNC—china network communicate　中国网通

COMSAT—Communications Satellite Corporation　美国通信卫星公司

CTT—China TieTong 中国铁通

CTIA—Cellular Telecommunications Industry Association　蜂窝电信工业协会

IEC—international electrotechnical commission　国际电工委员会

ICNIRP—international commission on non-ionizing radiation protection　国际非电离辐射保护委员会

ITU-T—international telecommunication union-telecommunication standardization sector
　　国际电信联盟电信标准局

ETSI—European Telecommunications Standards Institute　欧洲电信标准学会

FCC—Federal Communications Commission，（美国）联邦通信委员会

ICSA—International Computer Security Association　国际计算机安全协会

IEC—International Electrotechnical Commission　国际电工委员会

IEE—Institute of Electrical Engineers　电气工程师学会（英国）

IEEE—Institute of Electrical and Electronics Engineers　电气和电子工程师学会（美国）

ITU—International Telecommunication Union　国际电信联盟

ITU-R—ITU-Radio communications sector　国际电信联盟无线电通信组

ITU-T—ITU-Telecommunication standardization sector　国际电信联盟电信标准化组

ITSO—International Telecommunications Satellite Organization　国际电信卫星组织

ISO—International Standardization Organization　国际标准化组织

INTELSAT—INternational TELecommunication SATellite　国际通信卫星（组织）

IMTC— Internatinal Multimedia Television Committee　国际多媒体电视委员会

IFIP—International Federation for Information Processing　国际信息处理联合会

TDK—Tokyo Dengikagaku Kogyo K.K　东京电气化学工业株式会社

TIA (Telecommunications Industry Association)　（美国）电讯工业协会

附录 C　习题答案

练 习 1

1.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）simplex communication system

（2）full-duplex communication

（3）synchronous communication

（4）asynchronous communication

（5）point-to-point communication

（6）serial communication

（7）parallel communication

（8）telecommunication

（9）Cable television

（10）Photoelectricity processing technology

（11）radio astronomy

（12）satellite communication

（13）large-scale-integrated circuit (LSI）

（14）Very-large-scale-integrated circuit (VLSI）

（15）Nyquist's Theorem

（16）Non-distortion Coding Principle

（17）Error-correction Coding

1.3.2 仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (electromagnetic);　(transmit);　(telecommunication）
2. (simplex);　(full-duplex）
3. (serial);　(parallel）
4. (synchronous);　(asynchronous）
5. (digital); (analog）

1.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1. （ T ）;　　2. （ T ）;　　3. （ F ）;　　4. （ F ）;　　5. （ F ）;

6. （ T ）；　7. （ F ）;　　8. （ T ）。

练 习 2

2.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）analog communication system

（2）digital communication system

（3）Analog/Digital (A/D）conversion

（4）baseband waveform

（5）the rate of the code (the code rate）

（6）adjacent channel interference
（7）baseband signal
（8）phase modulation
（9）voltage-controlled oscillators (VCOs）
（10）AB amplifiers
（11）p-n junction diode
（12）single sideband signal
（13）signal-to-noise ratio
（14）Taylor series
（15）frequency-to-amplitude converter
（16）intermediate frequency (IF）
（17）zero-crossing detector
（18）phase comparator
（19）feedback loop
（20）FM threshold

2.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (digital);　(analog);
2. (amplitude）
3. (Sampling);　(Quantizing);　(Coding);
4. （source encoding);　(data compression);
5. （redundancy);　(noise);　(interference）
6. (linear);（nonlinear）
7. (distort); (envelope）
8. (efficiency); (poor); (adjacent channel interference); (loss）
9. (amplitude);　(baseband signal）
10. (frequency modulation); (phase modulation）
11. (direct); (indirect）
12. (slope detection); (zero-crossing); (phase locked discrimination); (quadrature detection）
13. (envelope);　(amplitude）
14. (frequency band）

2.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1.　（ F ）；　2.　（ T ）；　3.　（ F ）；　4.（ T ）；
5.　（ T ）；　6.　（ F ）；　7.　(F ）；　8.　(T ）；
9.　(F ）；　10.　(T ）；　11.　(T ）；　12.　(F ）；
13.　(T ）；　14.　(T ）

练 习 3

3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）Huffman source encoding algorithm
（2）analog source

（3）DMS (Discrete Memoryless Source）

（4）channel capacity

（5）code words

（6）fixed-weight code / constant-weight code

（7）Cyclic block code

（8）error correction capability

（9）discrete sources

（10）Shannon-Fano source encoding algorithm

（11）sampling theorem

（12）Convolutional Codes

（13）probabilistic approach

（14）non-binary code

（15）automatic repeat-request (ARQ）

（16）Hamming distance

3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (analog);　(discrete）

2. (discrete);　(digitally);　(encoder);　(sequence)

3. ($H(X)$);　(average);　(letter);　($H(X)$)

4. (efficient);　(symbol);　(block);　(close to)

5. (Shannon-Fano);　(Huffman);　(Huffman)

6. ($\log_2 L$);　(is);　($R=\log_2 L+1$);

7. (upper limits);　(reliable);　(Shannon)

8. (reliable);　($R<C$);　($R>C$);　(zero)

9. (number);　(n);　(binary);　(bits);　(non-binary)

10. (b);　($b \cdot N$)

11. (2^n);　(n);　(k/n);

12. (correctable);　(t);　(t);　($e_c \leqslant e_d$）

3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1. （ T ）；　2. （ T ）；　3. （ F ）；　4. （ F ）；　5. （ F ）；

6. （ T ）；　7. （ F ）；　8. （ F ）；　9. （ T ）；　10.（ F ）；

11.（ T ）；　12.（ T ）；　13.（ F ）。

练 习 4

4.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）Multiple access

（2）wireless communication network

（3）multi-user communication system

（4）frequency non-overlapping sub-channel

（5）frequency-division multiple access (FDMA）

（6）time slot

（7）mobile cellular communication system

（8）code-division multiple access (CDMA）

（9）spread spectrum signal

（10）Spread-spectrum radio communications

（11）bandwidth expansion factor

（12）low-probability-of intercept (LPI ）

（13）Frequency-Hopped (FH）spread spectrum

（14）satellite channels

（15）multi-path propagation

（16）background noise

（17）multiple-access communication systems

（18）fast-hopped signal

4.3.2 仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (non-overlapping); (frequency-division multiple access); (multiple)
2. (sub-channels); (non-overlapping); (time-division multiple access); (TDMA)
3. (inefficient); (available); (time); (carry)
4. (alternative); (sub-channel); (spread);
5. (unique); (cross-correlation); (pseudo-random);
6. (time); (frequency); (spread-spectrum multiple access）or (SSMA).
7. (greater); (rate); (W/R); (redundancy)
8. (PN); (demodulate);
9. (common); (pseudo-random）or (PN); (corresponding); (common)
10. (power); (power); (probability); (detected)
11. (intended); (pseudo-random）or (PN); (do not)
12. (encoder); (demodulator); (pseudo-random pattern）or (PN pattern); (pseudo-random）or (PN); (modulator)
13. (gain); (gain); (channel); (distinct);
14. (orthogonality); (cross-correlation); (orthogonal); (interference)
15. (CDMA); (DS); (frequency slot); (1/W); (are not）
16. (time); (pseudo-randomly); (RT); (RT/R_c); (RT/R_c); ($1000R/R_c$); ($1000R/R_c$ ）
17. (combining); (FH); (hop); (DS); (hop);

4.3.3 判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1. （ F ）；	2. （ F ）；	3. （ T ）；	4. （ F ）；	5. （ T ）；
6. （ T ）；	7. （ F ）；	8. （ F ）；	9. （ F ）；	10. （ T ）；
11. （ T ）；	12. （ F ）；	13. （ T ）；	14.（ T ）；	15.（ F ）。

练 习 5

5.3.1 请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）Symbol synchronization

（2）phase-locked loop (PLL）

（3）carrier synchronization

（4）Costas loop

（5）double-sideband suppressed carrier (DSB）signal

（6）phase ambiguity

（7）Squaring loop

（8）self-synchronization

（9）very low frequency band (VLF）

（10）Carrier recovery

（11）symbol rate

（12）timing recovery

（13）inter-symbol interference (ISI）

（14）eye pattern

（15）partial-response signals

（16）Ⅰpartial-response systems

（17）matched filter

5.3.2 仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (Symbol); (carrier)
2. (pilot); (local); (phase); (phase-locked loop）or (PLL); (narrow); (presence)
3. (symbol); (sampling); (clock); (symbol); (timing)
4. (frequency); (symbol interval); (timing)
5. (master); (compensate); (relative); (delay)
6. (extracted); (receiver); (self-synchronization)
7. (inter-symbol interference）or (ISI); (adjacent)
8. (band-limited); (filtering); (encoding)
9. (eye pattern)

5.3.3 判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1.（ F ）; 2. （ T ）; 3. （ F ）; 4. （ F ）; 5. （ T ）

6.（ T ）; 7. （ F ）; 8. （ T ）; 9. （ T ）; 10.（ T ）

11.（ F ）; 12. （ T ）; 13.（ T ）; 14.（ F ）。

练 习 6

6.3.1 请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）IMEI (International Mobile Subscriber Identity)

（2）HLR —— Home Location Register

（3）VLR ——Visitor Location Register

（4）MSC ——Mobile Switching Centre

（5）base station subsystem

（6）Global System for Mobile Communications (GSM)

（7）Personal Communication Services (PCS)

（8）switching subsystem

（9）hand-held cellular telephone

（10）BSC ---- Base Station Controller

（11）AuC ---- Authentication Centre

（12）GMSC ---- Gateway Mobile Switching Centre

（13）MSRN (Mobile Station Roaming Number)

（14）physical channels

（15）logical channels

（16）Code Excited Linear Prediction (CELP)

（17）Global Position System (GPS)

6.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (Global System for Mobile Communication); (standard); (GSM900); (DCS1800); (protocols);
2. (0.8); (20); (35); (500); (switching); (base station）or (BS);
3. (MS); (Base Transceiver Station); (reception); (air）
4. (subscriber); (permanent); (MSC); (database); (permanent); (location);
5. (turning-on); (register); (initialization); (judges);
6. (American National Standard Institute）or (ANSI); (IS-95); (limitation); (validity); (3)
7. (narrow CDMA）or (N-CDMA); (IS-95B); (64);
8. (W-CDMA); (TD-SCDMA); (2); （144)
9. (CDMA/FDD); (link); (PN); （1.23)

6.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1.（ T ）；　2.（ F ）；　3.（ F ）；　4.（ T ）；　5.（ T ）；

6.（ F ）；　7.（ T ）；　8.（ F ）；　9.（ F ）；　10.（ T ）；

11.（ F ）；　12.（ F ）；　13.（ T ）；　14.（ F ）；　15.（ F ）；

16.（ F ）。

练 习 7

7.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）fax

（2）modem

（3）Integrated Service Digital Network (ISDN）

（4）Internet browser

（5）trunk line

（6）call forwarding

（7）Class 5 switch

（8）service logic

（9）call

（10）signaling

（11）in-band

（12）out-of-band

（13）Service Switching Point (SSP）

（14）Intelligent Network (IN）

7.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

（1）(medium)；(larger)

（2）(Telephone sets)；(fax machines)；(modems)

（3）(PBX)

（4）(PBX)；(IP)；(packet)；(IP)

（5）(Converged)；(voice)

（6）(content)；(route)；(destination)

（7）(communication)

（8）(Public Switched Telephone Network)

（9）(call)；(packet)

（10）(call)；(network)；(maintenance)

（11）(signalling transfer point)

（12）(call)；(Intelligent Network)；(control)

7.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1. （ T ）；　2. （ T ）；　3. （ F ）；　4. （ T ）；　5. （ T ）；
6. （ F ）；　7. （ T ）；　8. （T ）；　9. （ T ）；　10.（ F ）；
11. （ F ）；　12. （ T ）；　13. （ T ）；　14. （ T ）；　15. （ F）；
16. （ F ）；　17. （ T ）；　18. （ F ）；　19. （ F ）。

练 习 8

8.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）cell

（2）routing

（3）roaming

（4）chip rate

（5）bandwidth

（6）wireless access

（7）uplink

（8）Personal Communication Services (PCS）

（9）time slot

（10）paging system

（11）mobile station

8.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (third-generation)
2. (analogue)
3. (digital)
4. (ITU)
5. (Third Generation Partnership Project 2)
6. (UMTS)；　（1920）

7. (W-CDMA)；(GSM)；(GSM)

8. (Personal Communication Services)

9. (Time Division-Synchronous Code Division Multiple Access)

10. （1.6); (smart antennas); (spatial filtering); (joint detection)

8.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1. （ F ）；　2. （ T ）；　3. （ F ）；　4. （ T ）；

5. （ F ）；　6. （ F ）；　7. （ T ）；　8. （ F ）；

9. （ F ）;　10. （ F ）；　11. （ F ）；　12.（ T ）。

练 习 9

9.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）Internet of Things (IoT）

（2）Radio-frequency identification (RFID）

（3）radio tag

（4）barcodes

（5）2D-codes

（6）smart city

（7）Information and Communication Technologies (ICTs）

（8）Wireless sensor networks

（9）distributed network

（10）automated control

（11）graphical user interface（GUI）

9.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

（1）identifiable，Internet-like

（2）Radio-frequency identification

（3）radio tags，barcodes，2D-codes

（4）minuscule identifying

（5）RFID，QR Codes

（6）strategic ，Information and Communication Technologies (ICTs）

（7）distinguish，intelligent

（8）wirelessly，real-time

（9）choreographed

（10）touch screen，back-end

9.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1. （ T ）;　2. （ F ）;　3. （ T ）;　4. （ T ）;　5. （ F ）;

6. （ F ）;　7. （ F ）;　8. （ T ）;　9. （ T ）。

练 习 10

10.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）streamed multimedia

（2）global roaming

（3）High Speed Downlink Packet Access（HSDPA）
（4）packet switched network
（5）circuit switched network
（6）spread spectrum transmission
（7）Multi-input Multi-output（MIMO）
（8）base station
（9）licensed spectrum
（10）multiple antennas
10.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。
1. (fourth-generation);（IP);（streamed）
2. (LTE);(WiMAX)
3. (spectrally);(dynamically);（cel）
4. (IMT-2000); (peak)
5. (HSPA+)
6. (LTE)
7. (3GPP);(release)
8. (circuit); (all-IP)
9. (transmitter); (receiver)
10. (Wi-Fi); (access)
10.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。
1.（ T ）　2.（ F ）　3.（ F ）　4.（ T ）　5.（ T ）
6.（ F ）　7.（ T ）　8.（ T ）　9.（ F ）　10.（ F ）
11.（ T ）

练 习 11

11.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。
（1）circuit-switched network
（2）router
（3）buffer
（4）media access control (MAC）
（5）relaying node
（6）Public Switched Telephone Network (PSTN）
（7）system-level
（8）packet-switched network
（9）additional POH
（10）propagation delay
（11）Ethernet-based LAN
（12）LAN-based IP network
（13）Ethernet network
（14）Ethernet frame

（15）physical connecting

（16）transparent path

11.3.2 仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (transmission)
2. (dedicated)
3. (analog)；(digital)
4. (packets)
5. (LAN)；(WAN)；(telephone)；(packet)
6. (spectral congestion);(user capacity)
7. (increased); (capacity); (spectrum)
8. (allocation); (reuse）
9. (frequency reuse); (frequency planning)
10. (transceiver); (antenna); (control circuitry)
11. (common air interface)
12. (roaming)

11.3.3 判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1.（ T ）； 2. （ T ）； 3. （ F ）； 4. （ T ）； 5. （ F ）；

6. （ T ）； 7. （ F ） 8. （ F ） 9. （ F ） 10. （ T ）。

练 习 12

12.3.1 请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）refractive index

（2）angle of incidence

（3）critical angle

（4）core

（5）cladding

（6）modal dispersion

（7）Single-mode fiber

（8）Multi-mode fiber

（9）Wireless Local Loop (WLL）

（10）Local Multipoint Distribution Service （LMDS）

12.3.2 仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (high data rate capabilities); (noise rejection); (electrical isolation)
2. (interface); (speeds)
3. (refract); (surface); (reflect)
4. (1); (1); (1.33); (1.5）
5. (critical angle); (cone);
6. (step-index); (graded-index); (singlemode)
7. (Wireless Local Loop)
8. (Local Multipoint Distribution Service)

9. (150); (300); (1300)

12.3.3 判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1. （ F ）； 2. （ T ）； 3. （ F ）； 4. （ T ）； 5. （ F ）；
6. （ T ）； 7. （ T ）； 8. （ F ）； 9. （ T ）； 10.（ F ）；
11.（ T ）； 12. （ T ）.

练 习 13

13.3.1 请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）asynchronous transfer mode (ATM）
（2）quality-of-service
（3）Liner Predictive Coding (LPC）
（4）packet
（5）binary
（6）mechanism
（7）algorithm
（8）Code Excited Linear Prediction (CELP）
（9）Bluetooth
（10）Personal Area Network(PAN）
（11）handoff
（12）soft handoff
（13）mobile assisted handoff (MAHO）

13.3.2 仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1. (Internet)；(Internet Protocol)
2. (packet）；(circuit）
3. (packets）；(routing)
4. (compressed)
5. (compressor/de-compressor)
6. (frequency）；(speech)
7. (SIP）；(H.323)
8. (end-to-end）；(real-time)
9. (8000 ）；(number）
10. (2.4); (TDD)
11. (1MHz); (1600)
12. (forward error control coding); (automatic repeat request)

13.3.3 判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1.（ F ）； 2. （ T ）； 3. （ F ）； 4. （ T ）； 5. （ T ）；
6.（ F ）； 7. （ T ）； 8. （ F ）； 9. （ T ） 10. （ T ）
11.（ F ） 12. （ F ）； 13.（ T ）； 14. （ T ） 15. （ F ）

练 习 14

14.3.1 请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）instruction set

（2）set-top box

（3）microcontroller

（4）semiconductor

（5）transistor

（6）SoC

（7）GPU

（8）radio baseband

（9）digital signal processing

（10）peripheral

（11）firmware

（12）VLIW

14.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。

1.(RISC)，(ARM)

2. (Advanced), (Acorn)

3. (costs), (heat), (power usage), (battery)

4. (chip)，(RAM), (GPUs),　(radio basebands)

5. (processor)

6. (control)

7. (microcontrollers)，(digital signal processors)

8. (firmware), (read-only)，(Flash)

9. (microprocessors)，(microcontrollers), (peripherals)

10. (subsystem)

14.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1.（ F ）；2.（ T ）；3.（ T ）；4.（ F ）；5.　（ T ）；

6.（ F ）；7.（ T ）；8.（ F ）；9.（ T ）；10.（ F　）；11.（ T ）。

练 习 15

15.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。

（1）Broadband integrated services digital network (B-ISDN)

（2）Carrier to Noise Ratio (CNR)

（3）Basic Rate Access (BR)

（4）Signaling System No. 7 (SS7)

（5）Regeneration, Reshaping, Retiming

（6）Synchronous Digital Hierarchy (SDH)

(7）Synchronous Optical Network (SONET)

(8）Visual Phone

（9）Computer Telecommunication Integration (CTI)

（10）Information Communication Technology (ICT)

（11）Wavelength Division Multiplexing (WDM)

（12）Frame Relay (FR)
（13）Electromagnetic Interference
（14）Electro Magnetic Compatibility (EMC)
（15）Time Division-Synchronous Code Division Multiple Access (TD-SCDMA)
（16）Orthogonal Frequency Division Multiplexing (OFDM)
（17）Multi-Carrier Modulation (MCM)
（18）Ultra Wide Band (UWB)
（19）Multiple-Input Multiple-output (MIMO)
（20）Cordless Telephone (CT)
（21）Diversity Receiver (DR)
（22）Adaptive Differential Pulse Code Modulation (ADPCM)
（23）Automatic Transfer Power Control (ATPC)
（24）Internet Protocol Television (IPTV)
（25）Handoff
（26）Quadrature Amplitude Modulation (MQAM)
（27）General Packet Radio System (GPRS)
（28）Public Switched Telephone Network (PSTN)
（29）Universal Personal Telecommunications Number(UPTN)
（30）Digital Data Network (DDN)
（31）Asynchronous Transfer Mode (ATM)
（32）Home Page
（33）Website
（34）Content Delivery Network
（35）Supplementary services
（36）Network Interface Card (NIC)
（37）Virtual Network Operator
（38）Network Management
（39）Network Paralysis
（40）Hit Count
（41）Security Association (SA)
（42）Dynamic Random-Access Memory (DRAM)
（43）Bearer Service
（44）Service Level Agreement
（45）Conversational Service
（46）Local telephone service
（47）Long distance telephone service
（48）Ampl itude Shift Keying (ASK)
（49）Bluetooth
（50）Trellis Coded Modulation (TCM)
（51）Personal Hand phone System

（52）Community Antenna Television (CATV)
（53）electronic program guide
15.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。
（1）(generation）, (NGN）
（2）(TD-SCDMA); (3G); (China）
（3）(Telecommunication); (interactive); (added）
（5）(information);　(communication）
（6）(OFDM);　(orthogonal);　(parallel);　(Interference）
（8）（100);　(200）
（11）(integration);　(communication);
（12）(General Packet Radio System);　（115);　(on-line);　(connection）
（13）(interactive);　(Internet）
15.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。
1.（ T ）；　2.（ T ）；　3.（ F ）；4.（ T ）；　5.（ F ）；
6.（ F ）；　7.（ T ）；　8.（ T ）；　9.（ F ）；　10.（ T ）；
11.（ F ）；12.（ T ）；　13.（ T ）；14.（ F ）；15.（ T ）；
16.（ T ）；17.（ T ）；　18.（ F ）；19.（ F ）；　20.（ T ）；　21.（ F ）。

练 习 16

16.3.1　请将下列中文词汇译作英文，有英文标准缩写形式的也一并写出。
（1）engine control unit
（2）internal combustion engine
（3）look-up table
（4）idle speed
（5）printed circuit board
（6）Engine Management System
（7）Controller Area Network
（8）drivetrain
（9）gyroscope
（10）Infotainment systems
（11）multipath
（12）transmission loss
16.3.2　仔细阅读下列各句，选择适当的词汇、短语、缩写或数字填空。
1. (ECUs)，(engine)，（transmission)
2. (powertrain)，(actuators)，(combustion)
3. (air/fuel mixture), (ignition timing), (idle speed)
4. (microprocessor)，(sensors)，(hardware），(software).
5. (Engine Management System).
6. (transmission), (anti-skid brake)，(anti-theft).
7. (drivetrain),　(gyroscope)，(accelerometer).

8. (electric)，(traction)，(propulsion).

9. (fossil).

10. (locomotion)，(overhead lines)，(loss).

16.3.3　判断下列说法正确与否，并将答案填写在每题序号后面的括号内。

1.（ T ）；2.（ T ）；3.（ T ）；4.（ F ）；　5.（ T ）；

6.（ F ）；7.（ T ）；8.（ F ）；9.（ F ）；　10.（ T ）。

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